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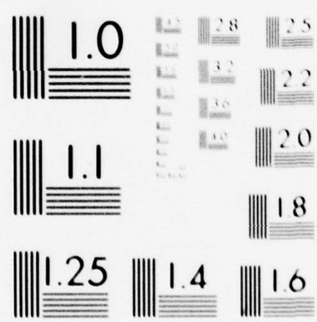

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MASSACHUSETTS INSTITUTE OF TECHNOLOGY
LINCOLN LABORATORY

NETWORK SPEECH PROCESSING PROGRAM

ANNUAL REPORT
TO THE
DEFENSE COMMUNICATIONS AGENCY

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ABSTRACT

This report documents work performed during FY 1978 on the DCA-sponsored Network Speech Processing Program. Three areas of work are reported: (1) a voice/data integration study investigating the effectiveness of combined circuit and packet multiplexing techniques; (2) a study of Demand-Assignment Multiple Access (DAMA) schemes for future integrated satellite networks; and (3) planning of experiments for an Experimental Integrated Switched Network (EISN) test bed.

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NETWORK SPEECH PROCESSING

I. INTRODUCTION AND SUMMARY

This report documents work at Lincoln Laboratory during FY 1978 on the DCA-sponsored Network Speech Processing Program which consists of three major tasks: (a) a voice/data integration study investigating the effectiveness of hybrid (combined circuit and packet) multiplexing techniques; (b) a study of Demand-Assignment Multiple Access (DAMA) schemes for future integrated satellite networks; (c) planning of experiments for an Experimental Integrated Switched Network (EISN) test-bed being developed under joint DARPA/DCA sponsorship. An important purpose of the two systems studies is to assist in the identification and definition of significant experiments for the network test bed.

Section II deals with the voice/data integration study. Two hybrid multiplexing techniques, Slotted Envelope Network (SENET) and SENET Virtual Circuit (SVC), are considered. An investigation of the effects of various forms of flow control in a SENET multiplexer is reported. This work was an extension of a simulation and analysis effort described in the previous Annual Report,[1] in which a basic simulation model was developed and the need for flow control was identified. The best flow-control results were obtained with limitation of the data buffer combined with data-queue-dependent voice-rate control. The results of simulation of an SVC system, which differs from SENET in that Speech Activity Detection (SAD) is exploited so that voice users transmit only during talkspurts, are described. Comparisons of the performance of data traffic for SENET, SVC, and Packetized Virtual Circuit (PVC) are presented. Performance is shown to depend more on the fundamental characteristics of voice and data traffic than on the details of the multiplexer scheme.

The DAMA study is the subject of Sec. III. The focus is on the problem of taking full advantage of SAD in a situation where a large number of ground stations, with a relatively small number of voice users at each ground station, share a broadcast satellite channel. A prediction scheme, which has the potential for effective anticipation of speaker activity at a ground station in time to obtain the required capacity through a reservation request on the satellite channel, is derived and verified for measured talkspurt/silence characteristics. The effect of buffering of speech at each ground station is investigated, and a fundamental trade-off is demonstrated between the amount of delay that is allowed and the efficiency or "TASI (Time-Assigned Speech Interpolation) advantage" associated with the exploitation of SAD.

Section IV consists of a preliminary experiment plan for EISN. A development plan for the network test bed is presented, with descriptions of major network subsystems. Advanced systems experiments are defined and categorized. Seven major experimental areas are identified, and objectives and performance measures are described for each class of experiments.

Since the development of the network test bed and the experiment planning effort are jointly supported by DCA and DARPA, a similar description of the preliminary experiment plan appears in the 30 September 1978 Semiannual Technical Summary for the DARPA Wideband Integrated Voice/Data Technology Program. Some of the experiment areas discussed are of more-immediate concern to one or another of the two sponsoring agencies. For example, the DCA is particularly interested in experiments related to satellite DAMA and to switching techniques for voice/data integration. However, the problem areas are so interrelated and complementary that all are of interest to both agencies; therefore, descriptions of all experiment areas are included in the reports to both agencies. The experiment plan is expected to grow and become more detailed as the program proceeds. In fact, a more-detailed version of the preliminary experiment plan is being submitted to DCA and DARPA under separate cover. This separate version is more specific with respect to schedules and requirements, and is intended to serve as a working document for coordination among the experimental network program participants and sponsors.

The previous Annual Report[1] included work in the Secure Voice Conferencing area. Work in the DCA Secure Voice Conferencing program during FY 78 is reported under separate cover.

II. VOICE/DATA INTEGRATION STUDY

This section reports on efforts during FY 78 in the study of hybrid techniques for voice/data integration. Two hybrid multiplexing techniques – Slotted Envelope Network (SENET)[2] and SENET Virtual Circuit (SVC) – are considered. An investigation of the effects of various forms of flow control in a SENET multiplexer is reported. This work was an extension of a simulation and analysis effort described in the previous Annual Report,[1] in which a basic simulation model was developed and the need for flow control was identified. The best flow-control results were obtained with limitation of the data buffer combined with data-queue-dependent voice-rate control. The results of simulation of an SVC system, which differs from SENET in that Speech Activity Detection (SAD) is exploited so that voice users transmit only during talkspurts, are described. Comparisons of the performance of data traffic for SENET, SVC, and Packetized Virtual Circuit (PVC) are presented. Performance is shown to depend more on the fundamental characteristics of voice and data traffic than on the details of the multiplexer scheme.

A. INVESTIGATION OF FLOW-CONTROL TECHNIQUES FOR SENET

In the previous Annual Report,[1] the results of a SENET voice/data multiplexer were presented. The conclusion was that attempts to achieve high channel utilization led to data-packet queues so large as to overflow any reasonable amount of storage in the multiplexer. Even assuming infinite storage, the mean data delays were large because of the buffer buildup during periods when voice channel occupancy was high. The need for a flow-control mechanism became apparent, and two types of flow control have been investigated: (1) voice-rate control, where voice coders are assumed to be capable of operating at a variety of rates and a new voice call is assigned (at dial-up) one of several available bit rates, based on the current voice utilization and/or data-queue size; and (2) data-flow control, where a fixed limit is imposed on the size of the data queue. A unified description of both the basic SENET analysis and the flow-control investigation is presented in Ref. 3. This report focuses on the study of flow-control techniques.

Before discussing the flow-control investigations, it is useful to briefly review the SENET multiplexing scheme.

SENET is a Time Division Multiplex scheme which is a hybrid of circuit- and packet-switching techniques. Time slices of fixed duration, a frame period, are partitioned and allocated to the transmission of circuit-switched digital voice traffic and packet-switched data. The basic SENET frame structure is illustrated in Fig. II-1. Voice traffic is allowed to occupy

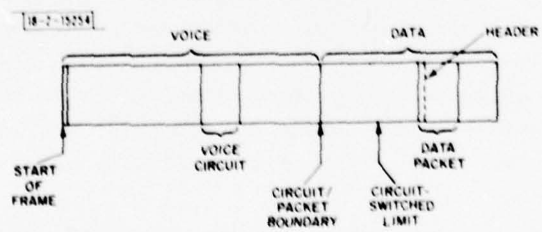


Fig. II-1. Frame structure for SENET hybrid multiplexing scheme.

up to a specified fraction of the frame, as specified by the circuit-switched limit indicated in the figure. The remaining portion of the frame is reserved for data packets. The circuit-switched limit must be selected to maintain a specified blocking probability for voice. Generally, some of the total maximum capacity for voice will be free due to statistical fluctuations in the voice traffic. This variable capacity for data, which is indicated in Fig. II-1 as the region between the circuit/packet boundary and the circuit-switched limit, may be utilized for transmission of data packets. In general, the voice slots (or circuits) can be of varying size depending on the bit rates of different voice users, and the size of data packets is also variable. The work reported in the previous Annual Report[1] assumed a multiplexer frame (taken to be of duration b = 10 msec) divided into S + N equal-capacity slots. The nominal bit rate for the voice coders was taken as 8 kbps, which is accommodated with an 80-bit slot size.

The flow-control investigations reported here include a set of simulations which have been run to gather statistics on multiplexer performance with different flow-control strategies. Generally, total channel capacity was fixed at 120 kbps, with voice capacity limited to 80 kbps (sufficient to accommodate N = 10 users at the nominal 8-kbps rate), and 40 kbps dedicated to data. (For comparison, one of the flow-control schemes was tested with a 600-kbps channel.) A movable boundary was employed to allow data packets to be sent in temporarily unused portions of the voice capacity. The usual telephone traffic model for voice calls was used, with a Poisson call arrival process and exponentially distributed holding times. For most of the experiments, the call arrival rate was set at $\lambda = 0.05\ \mathrm{sec}^{-1}$ and the mean holding time was taken as $1/\mu =$ 100 sec. This corresponds to 5 Erlangs of offered voice traffic. The maximum number of simultaneous voice calls was limited to 10, with blocked calls cleared. Data packets, with fixed lengths of 80 bits, were assumed to arrive in a Poisson process. The purpose of the simulations was to investigate system performance for data-packet arrival rates greater than 500/sec (or equivalently 5.0/frame), when a portion of the voice capacity must be utilized for data. For the cases where the data-packet arrival rate is less than 500/sec, no significant data queues build up. The performance measures of interest for data include average delay, average queue size, maximum queue size, and fraction of packets arriving when the data queue is full (when data flow control is employed). With voice-rate control, the distribution of callers among the available bit rates and the average voice bit rate assigned serve as performance measures for voice users.

1. Experiments with Voice-Rate Control but No Data-Flow Control

In the work reported on previously, all voice users were assumed to operate at 8 kbps. The voice channel utilization exhibited large variations around its mean, and data queues would build up during the peaks of voice utilization. The idea behind the voice-rate control techniques studies here was to cut down the peaks of voice channel utilization by assigning lower bit rates to callers who enter the system when utilization is high. Of course, this scheme assumes that each voice user has a flexible vocoder capable of a variety of rates. The presumption is that the assigned voice bit rate and the corresponding speech coder performance will vary depending on the traffic load in the system.

The results for four voice-rate control schemes are reported here. To define these schemes, the following notation is introduced:

R_V = sum of bit rates of all voice users active at a given time;

C_V = maximum channel capacity allocated for voice;

Q_d = number of packets currently in the data queue;

R_{new} = bit rate assigned to a new voice caller at dial-up.

Unless otherwise noted, C_V was fixed at 80 kbps. The first voice-rate control scheme that was tried is illustrated in Fig. II-2. New calls were assigned at 2, 4, 8, or 16 kbps as a function of current total voice channel utilization R_V/C_V, with lower rates assigned during higher utilization periods. The inclusion of a 16-kbps rate would provide better voice service for some users, but had the effect of filling in the valleys of voice utilization that were present when voice-rate

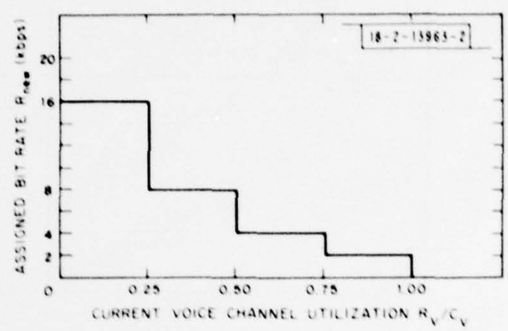


Fig. II-2. Example of voice-rate control based on current voice channel utilization. Bit rate assigned to a new user is plotted as a function of current total bit rate of all voice users.

control was not employed. The second scheme was similar to the first except that the voice rate of 16 kbps was not used, but 8 kbps was assigned to callers arriving when R_V/C_V was less than 0.25. The third approach used voice-rate control, combined with monitoring of the data queue. When the data queue was empty, voice rates were assigned as in Fig. II-2; however, voice callers arriving when the data queue was not empty were assigned a rate of 2 kbps. In the fourth technique, the rate of a new call was governed solely by the size of the data queue. If the queue was empty, a rate of 8 kbps was assigned. If the queue was non-empty but did not exceed 150 packets, 4 kbps was assigned. Otherwise, 2 kbps was used. The four approaches just outlined will be referred to below as schemes v1, v2, v3, and v4; v0 will denote the situation where no control is imposed. These voice-rate control schemes are summarized as follows:

$$\text{v0:}\quad R_{new} = \text{constant} = 8 \text{ kbps}$$

$$\text{v1:}\quad R_{new} = \begin{cases} 16 \text{ kbps} & R_V/C_V \leq 0.25 \\ 8 \text{ kbps} & 0.25 < R_V/C_V \leq 0.5 \\ 4 \text{ kbps} & 0.5 < R_V/C_V \leq 0.75 \\ 2 \text{ kbps} & 0.75 < R_V/C_V \leq 1.0 \end{cases}$$

$$\text{v2:}\quad R_{new} = \begin{cases} 8 \text{ kbps} & R_V/C_V \leq 0.5 \\ 4 \text{ kbps} & 0.5 < R_V/C_V \leq 0.75 \\ 2 \text{ kbps} & 0.75 < R_V/C_V \leq 1.0 \end{cases}$$

TABLE II-1

AVERAGE DATA-PACKET DELAY AS A FUNCTION OF DATA-PACKET ARRIVAL RATE FOR VOICE-RATE CONTROL SCHEMES
(v0 Through v4 as Described in Text)

Packet Arrival Rate (θb in packets/frame) \ Flow-Control Scheme	v0	v1	v2	v3	v4
	Average Delay (sec)				
6	0.18	0.0	0.0	0.0	0.0
7	1.18	0.0	0.0	0.0	0.12
8	3.08	1.19	0.14	0.33	0.30
9	11.6	39.0	1.6	2.55	0.63

TABLE II-2

AVERAGE BIT RATE ASSIGNED TO VOICE USERS FOR VOICE-RATE CONTROL SCHEMES v0 THROUGH v4

Packet Arrival Rate (θb in packets/frame) \ Flow-Control Scheme	v0	v1	v2	v3	v4
	Average Voice Bit Rate (kbps)				
6	8.0	5.7	6.6	6.0	7.9
7	8.0	5.7	6.6	6.0	7.6
8	8.0	5.7	6.6	5.5	7.2
9	8.0	5.7	6.6	4.8	6.5

$$v3: \; R_{new} = \begin{cases} \text{Assigned as in v1} & Q_d = 0 \\ 2 \text{ kbps} & Q_d \neq 0 \end{cases}$$

$$v4: \; R_{new} = \begin{cases} 8 \text{ kbps} & Q_d = 0 \\ 4 \text{ kbps} & 0 < Q_d \leq 150 \\ 2 \text{ kbps} & Q_d > 150 \end{cases}$$

Results on average packet delay as a function of data-packet arrival rate for schemes v0 through v4 are summarized in Table II-1. The average bit rate assigned to a voice user for each case is shown in Table II-2. The first scheme represents an improvement over no flow control for packet arrival rates $\Theta b = 6.0$, 7.0, and 8.0 packets/frame. (Recall that the frame duration $b = 10$ msec.) The price for this improvement was a drop in average voice bit rate from 8 to 5.7 kbps. However, the flow control worsened the situation when packet arrivals reached 9.0 per frame. When 16 kbps was eliminated as a voice rate (scheme v2), a general improvement in delay performance was realized. In addition, the average voice bit rate increased to 6.6 kbps because fewer users had to be assigned to the lower rates. Scheme v3 (the same as v1 except that some data-queue monitoring was introduced) represented a significant improvement over v1 in terms of data delay. Scheme v4, where voice-rate control was based strictly on data-queue size, was generally the most effective of the voice-rate control techniques.

However, none of the performance results shown in Table II-1 are satisfactory at the high packet arrival rates. Even for scheme v4, the delay of 0.63 sec at 9.0 packets/frame is unacceptable, and corresponds to an average queue size of 571 packets. Observed maximum queue sizes were significantly higher. The conclusion is that, although voice-rate control alone can enhance data performance, some form of data-flow control is necessary to keep delays and queue sizes within reasonable limits.

The emphasis here has been on the effectiveness of voice-rate control in reducing data-packet delay. However, the ability of voice users to operate at a variety of rates depending on traffic loads would enhance the voice-traffic handling capability of the system, independent of its effect on data traffic. For example, the assignment of lower bit rate to new voice users as the number of active calls increases could be utilized as a means for reducing the blocking probability for voice calls during busy periods.

2. Experiments with Combined Data-Flow Control and Voice-Flow Control

The first experiment including data-flow control involved limiting the data buffer to a fixed maximum size Q_{max}, but not including any voice-rate control. Q_{max} was (somewhat arbitrarily) set equal to 150 packets. Data packets were denied entry to the multiplexer when its queue was full. This represents added delay, since these packets have to be retransmitted to the multiplexer. If the packets enter the multiplexer directly from a user terminal, then the terminal would retransmit when no acknowledgment was received. Otherwise, a store-and-forward node feeding the multiplexer could handle the retransmission. The second experiment combined the same data-flow control procedure with the data-queue-independent voice-rate control employed in scheme v2 above. The third test combined limitation of the data buffer to 150 packets with data-queue-dependent voice-rate control similar to the method used in scheme v4 above. If the queue was empty, a rate of 8 kbps was assigned. If the queue was non-empty but did not exceed

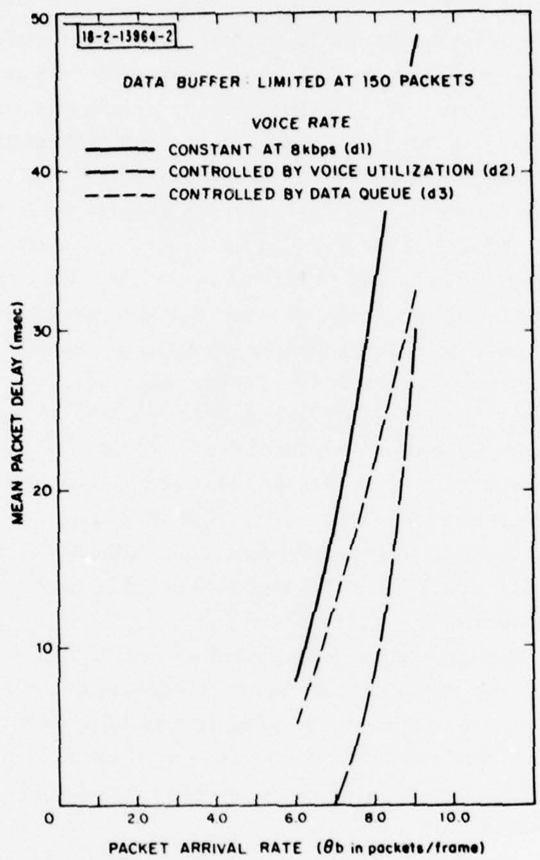


Fig. II-3. Average data-packet delay as a function of data-packet arrival rate for flow-control schemes d1 through d3.

75 packets, 4 kbps was assigned. Otherwise, 2 kbps was used. The three combined data/voice flow-control techniques just described will be referred to as d1, d2, and d3, and are summarized as follows:

d1, d2, d3: all impose fixed limit Q_{max} = 150 packets on data buffer size

d1: R_{new} = constant = 8 kbps

$$\text{d2:}\quad R_{new} = \begin{cases} 8 \text{ kbps} & R_V/C_V \leq 0.5 \\ 4 \text{ kbps} & 0.5 < R_V/C_V \leq 0.75 \\ 2 \text{ kbps} & 0.75 < R_V/C_V \leq 1.0 \end{cases}$$

$$\text{d3:}\quad R_{new} = \begin{cases} 8 \text{ kbps} & Q_d = 0 \\ 4 \text{ kbps} & 0 < Q_d \leq Q_{max}/2 \\ 2 \text{ kbps} & Q_{max}/2 < Q_d \leq Q_{max} \end{cases}$$

The average packet delays for the three cases are plotted in Fig. II-3. In all cases, the improvement in data-packet delay is dramatic compared with the situation where no data-flow control is imposed. For example, the worst case in Fig. II-3 is an average delay of 49 msec for scheme d1 at 9.0-packets/frame arrival rate. The corresponding best result without data-flow control is an average delay of 630 msec for v4 as shown in Table II-1.

Tables II-3 and II-4 show the average voice bit rates and the percentages of data packets arriving when the queue is full, as a function of packet arrival rate for the three data-flow control schemes. The tables include some measurements for packet arrival rates greater than 9.0 packets/frame. Table II-4 shows that the percentage of packets finding a full data queue is rather low in all cases. This implies that the additional delay seen by the user who must retransmit these packets also should be modest.

If schemes d1 through d3 are compared on the basis of data-packet delay alone, it is clear from Fig. II-3 that the best performance is achieved with d2, which includes voice-rate control independent of the data queue. However, the improvement in delay performance over d3 is counterbalanced by the fact that higher average voice bit rates were assigned in scheme d3 at all data-packet arrival rates tested.

Thus far, performance of the three data-flow control schemes has been displayed on the basis of packet arrival rate for convenient comparison with the voice-rate control schemes. In order to take into account the effects of packet discard due to data-flow control, and to provide a convenient means for comparing systems with different capacity, it is useful to present the results as a function of utilization. It is also desirable to focus on utilization of the variable capacity available to data due to fluctuations in voice traffic, rather than on utilization of fixed channel capacity dedicated to data. A correspondence between utilization of variable data capacity and packet arrival rate is shown in Table II-5. Variable data capacity is defined as the difference between the total capacity allotted for voice (80 kbps) and the average capacity actually utilized by voice. The average packet delays for the three schemes as a function of variable data capacity utilization are plotted in Fig. II-4. Notice that scheme d3 performs more favorably than scheme d2 in terms of packet delay as utilization exceeds 70 percent. Similarly, the

TABLE II-3

AVERAGE BIT RATE ASSIGNED TO VOICE USERS FOR FLOW-CONTROL SCHEMES d1 THROUGH d3

Packet Arrival Rate (θb in packets/frame) \ Flow-Control Scheme	d1	d2	d3
	Average Voice Bit Rate (kbps)		
6.0	8.0	6.6	7.9
7.0	8.0	6.6	7.7
8.0	8.0	6.6	7.3
9.0	8.0	6.6	6.8
9.5	8.0	6.6	6.4
10.0	–	6.6	6.0
10.5	–	6.6	5.5
11.0	–	–	4.5

TABLE II-4

PERCENTAGE OF PACKETS ARRIVING WHEN QUEUE IS FULL UNDER FLOW-CONTROL SCHEMES d1 THROUGH d3

Packet Arrival Rate (θb in packets/frame) \ Flow-Control Scheme	d1	d2	d3
	Packets Arriving When Data Queue Is Full (percent)		
6.0	0.3	0.0	0.1
7.0	1.1	0.0	0.4
8.0	2.4	0.2	0.7
9.0	4.2	1.4	1.1
9.5	6.8	2.8	1.5
10.0	–	4.6	1.8
10.5	–	6.9	2.5
11.0	–	–	3.3

TABLE II-5 PERCENT UTILIZATION OF VARIABLE CAPACITY AVAILABLE FOR DATA FOR FLOW-CONTROL SCHEMES d1 THROUGH d3			
Flow-Control Scheme → / Packet Arrival Rate (θb in packets/frame) ↓	d1	d2	d3
	Utilization of Variable Data Capacity (percent)		
600	19	18	19
700	39	37	37
800	55	55	55
900	70	71	70
950	78	78	76
1000	–	84	81
1050	–	88	87
1100	–	–	92

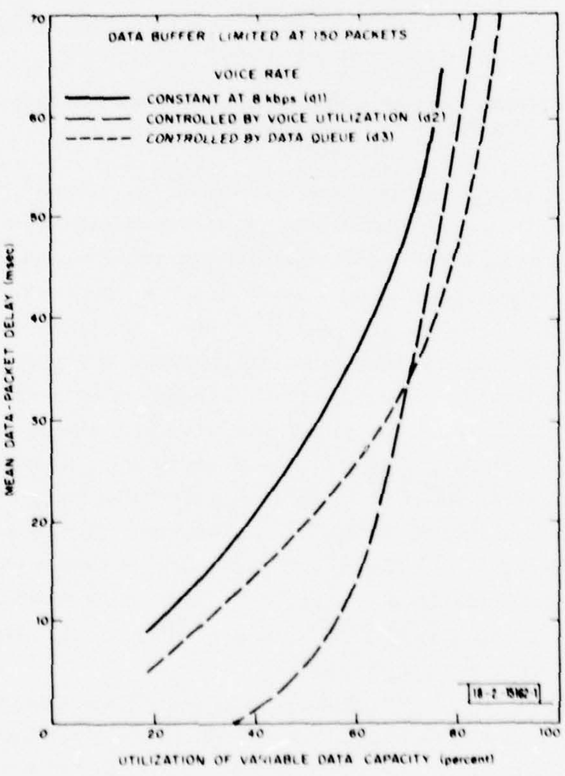


Fig. II-4. Mean packet delay as a function of utilization of variable data capacity for schemes d1 through d3.

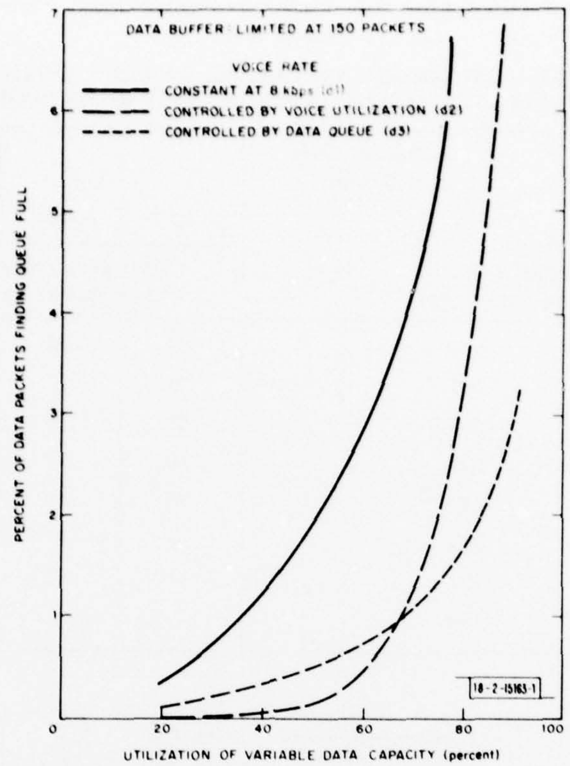


Fig. II-5. Percent of packets finding queue full as a function of utilization of variable data capacity for schemes d1 through d3.

percentages of data packets finding the queue full, which are plotted in Fig. II-5, show that d3 again has the best performance for higher utilization of variable data capacity. However, there is a price to be paid by using scheme d3. As utilization is increased by greater packet arrival rates, the data queue tends to remain more than half full during periods of higher voice utilization. Consequently, more calls are assigned at 2 kbps; and, lest the average voice bit rates shown in Table II-4 be misleading, it should be mentioned that for all packet arrival rates tested almost all calls were assigned at either 8 or 2 kbps using scheme d3. This is in contrast to scheme d2 where all calls were assigned at 8 or 4 kbps with a ratio of slightly less than two-to-one, regardless of packet arrival rate. If voice users could tolerate a greater overall incidence of 2-kbps voice communication with a better chance of being assigned at 8 kbps, it can be argued that scheme d3 is slightly better than scheme d2. The percentages of calls assigned at 8 kbps are plotted in Fig. II-6. The area of particular interest is that corresponding to a utilization of variable data capacity just over 70 percent, which is where scheme d3 shows improvement over scheme d2 in both Figs. II-5 and II-6. In general, comparable performances could be achieved with both schemes d2 and d3.

Scheme d3 was tested for a larger population of users by increasing the channel capacity from 120 to 600 kbps, with a voice capacity of 400 kbps and a dedicated data capacity of 200 kbps. The offered voice traffic for the larger population was set at 40 Erlangs. (With fifty 8-kbps channels of voice capacity and 40 Erlangs of traffic, the blocking probability computed on the basis

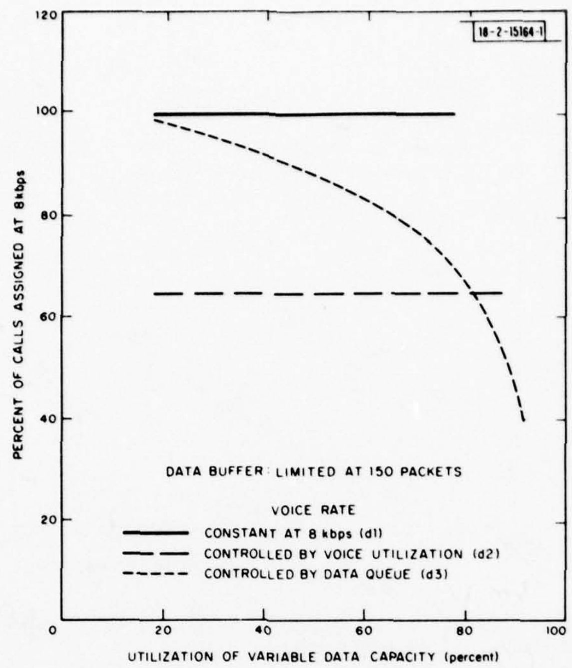


Fig. II-6. Percent of calls assigned at 8 kbps as a function of utilization of variable data capacity for schemes d1 through d3.

of fixed voice-rate operation at 8 kbps is $P_L = 0.0186$, approximately the same as the blocking probability for the 10-channel, 5-Erlang case considered up to now.) The performance of d3 with this larger population was compared with that of the smaller population considered above, where 80 kbps was allocated for voice and the offered voice traffic was 5 Erlangs. As shown in the plots of average data-packet delay, the percentages of packets finding the queue full, and the percentages of calls assigned at 8 kbps in Figs. II-7, II-8, and II-9, respectively, the performance was improved for the larger population. This can be attributed in part to the more-rapid fluctuation in voice utilization, which causes the peaks of voice utilization to be shorter in duration.

B. PERFORMANCE OF DATA TRAFFIC IN A SENET VIRTUAL CIRCUIT (SVC) MULTIPLEXER

The SENET simulation was expanded to model a SVC system in which Speech Activity Detection (SAD) is exploited. With SAD, voice-activated switching is employed, and voice users occupy their assigned slots only during talkspurts. Voice slots which are unused during periods of silence may be utilized for transmission of data packets. The multiplexer frame structure for SVC is shown in Fig. II-10. The organization is quite similar to the SENET frame shown in Fig. II-1, except for the inclusion of a speech-activity header. This header gives information as to which voice users are in talkspurt and therefore occupying their assigned slots during a particular frame period. Slots belonging to silence voice users are free for data usage. The circuit/packet boundary in Fig. II-10 moves forward and backward as calls enter and leave the system, under the constraint that calls will be blocked after the circuit-switched limit is

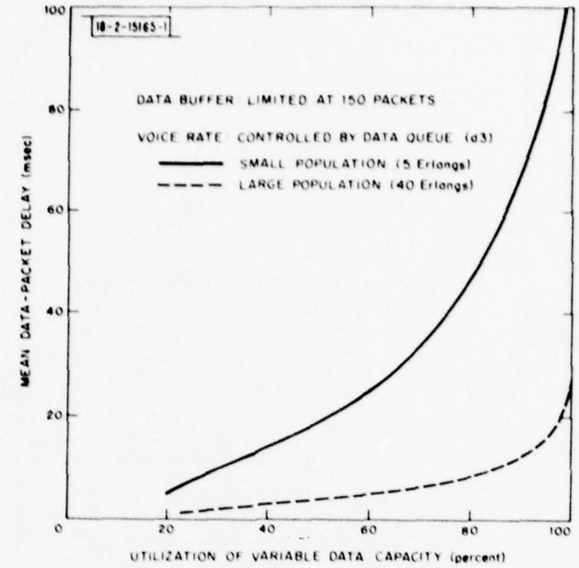


Fig. II-7. Mean packet delay for different user populations under flow-control scheme d3.

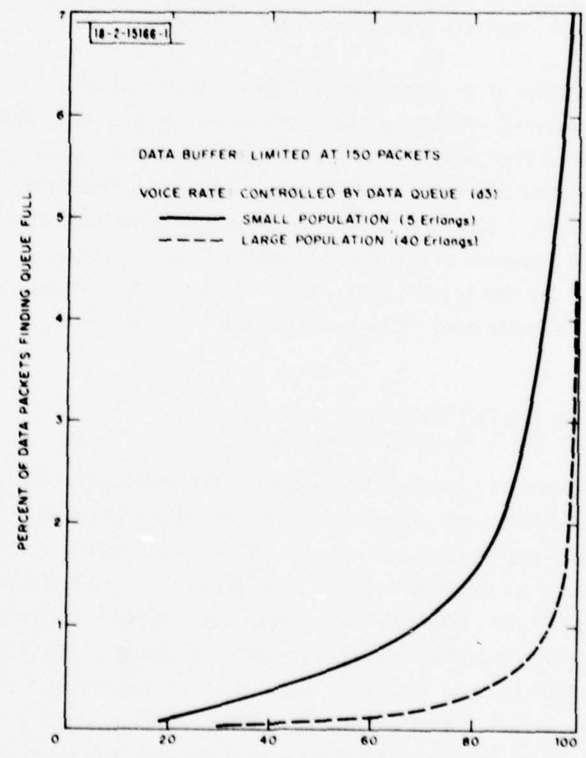


Fig. II-8. Percent of packets finding queue full for different user populations under flow-control scheme d3.

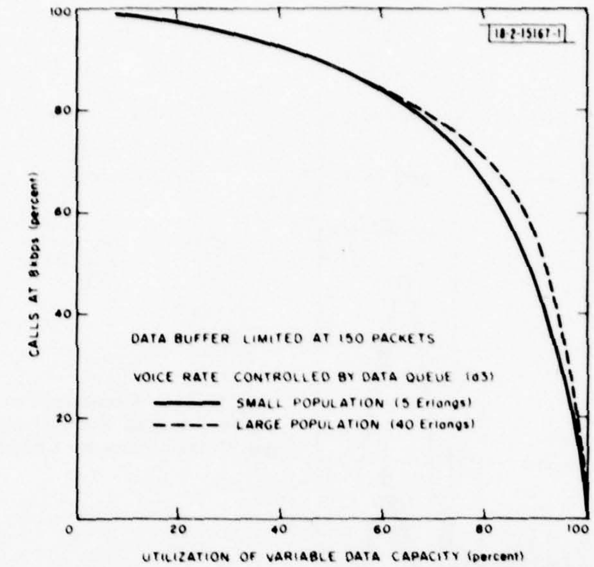
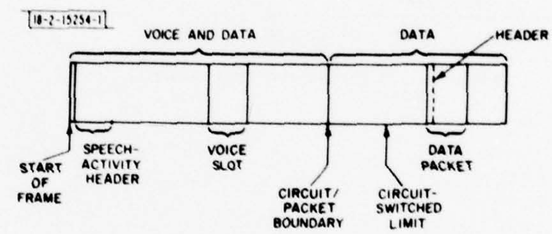


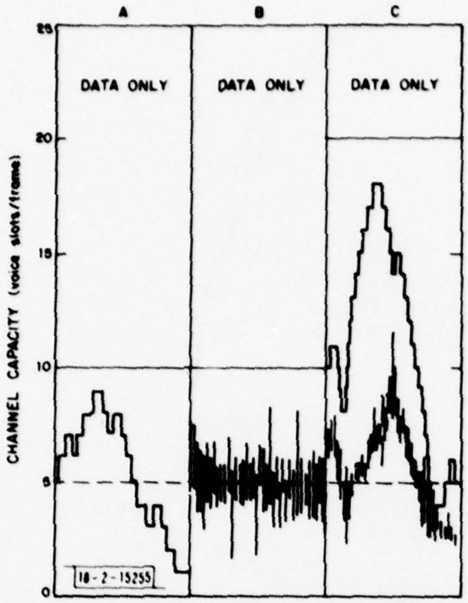
Fig. II-9. Percent of calls assigned at 8 kbps for different user populations under flow-control scheme d3.

Fig. II-10. Multiplexer frame structure for SVC.



exceeded. The SVC simulation allowed variation in voice utilization due to call initiations and terminations to be superposed on the talkspurt/silence variation.

The performance of data traffic for SENET was compared with the performance for SVC under the conditions of (1) a fixed number of calls and (2) a varying number of calls. The comparison was set up so that average voice utilization for the three cases of interest was the same. This was achieved by assuming a channel capacity of twenty-five 80-bit slots per 10-msec frame, or 200 kbps total. The average voice utilization in the three cases was arranged to be 5 slots/frame. For Case A (SENET), 10 slots were allowed for voice and 15 slots were dedicated for data, with a variable number of calls and an offered voice traffic of 5 Erlangs. For Case B (SVC), 10 slots were allowed for voice and 15 slots were dedicated to data, but the number of calls was fixed at 10, and speech transmission was assumed to occur only during talkspurts. A typical talker was assumed to be in talkspurt 50 percent of the time. For Case C (SVC), only 5 slots were dedicated to data, but call on/off variation was superposed on the talkspurt/silence variation with an offered voice load of 10 Erlangs. The three cases are illustrated in Fig. II-11. The three simulations were run for a packet arrival rate of 19 packets/frame, with the expectation that an average of 20 slots/frame would be available for data. For Cases A



CASE A: SENET, 5 ERLANGS OF VOICE TRAFFIC
CASE B: SVC, 10 TALKERS WITH SAD
CASE C: SVC, 10 ERLANGS OF VOICE TRAFFIC WITH SAD

Fig. II-11. Comparison of voice traffic variation and data channel availability for three cases as described in text.

TABLE II-6

COMPARISON OF DATA TRAFFIC PERFORMANCE FOR THE THREE CASES DEPICTED IN FIGURE II-11

Case	Maximum Data Queue	Mean Data Queue	Mean Delay (sec)	Utilization of Slots Available for Data (percent)
A	148,352.0	19,169.0	10.1	94.8
B	1,052.5	73.48	0.039	95.1
C	112,714.0	21,068.0	11.1	95.3

and C where an attempt was being made to exploit voice call variation for data packets, packet delay and queue were unacceptably large. This agreed with previous observations of data performance in SENET. Case B, where only talkspurt/silence variations were exploited for data traffic, resulted in much smaller values of mean data delay and queue size. Table II-6 shows the results of the simulations. Each entry represents an average over four runs per case, with each run representing 25,000 sec of real time. The results indicate that attempts to utilize as much as 80 percent of the average excess voice capacity due to fluctuations in the number of calls connected will result in large data delays whether or not SAD is included. The results for Case B are in agreement with previously obtained results for a Packetized Virtual Circuit (PVC) multiplexer, which also indicated that reasonably efficient use could be made of the voice capacity saved by not transmitting during silence.

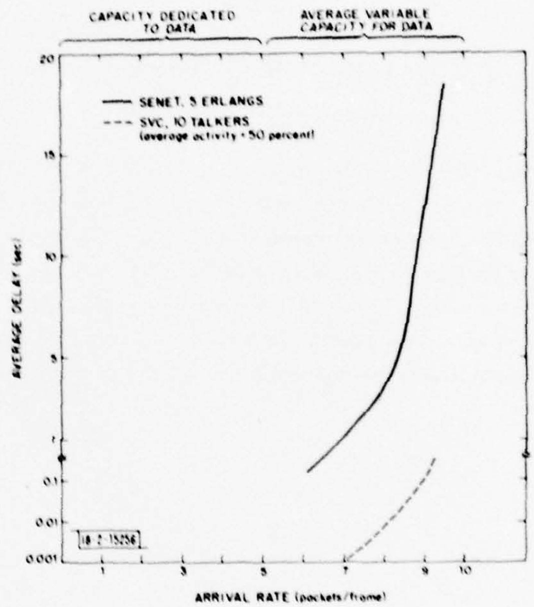


Fig. II-12. Mean packet delays for SENET with 5 Erlangs of voice traffic and SVC with 10 talkers.

The comparison just described is illustrated further by the results presented in Fig. II-12, which shows average delay as a function of packet arrival rate for a SENET system with 5 Erlangs of voice traffic and an SVC system with a fixed number of calls but with an average of 5 talkers in talkspurt. Here a total channel capacity of 15 voice slots was assumed, with 5 dedicated to data. The greater efficiency of data traffic in utilizing the rapid talkspurt/silence variations as opposed to the much slower call on/off variations is clearly illustrated.

The SVC simulation was run with parameters matching those previously[4] used in PVC simulation runs. A channel capacity equivalent to 72 one-way 16-kbps voice links was assumed. The number of connected calls was fixed at 85, no capacity was dedicated to data, and the mean durations of talkspurt and silence were set equal so that an average of 42.5 speakers was active. The total channel capacity was large enough so that all voice data were guaranteed to be transmitted. The SVC simulation was run to represent 2 min. of real time as had been done in the PVC experiment, and the results were smoothed over four runs per data arrival rate. The data arrival rate corresponding to an 80-percent utilization of the mean capacity for data

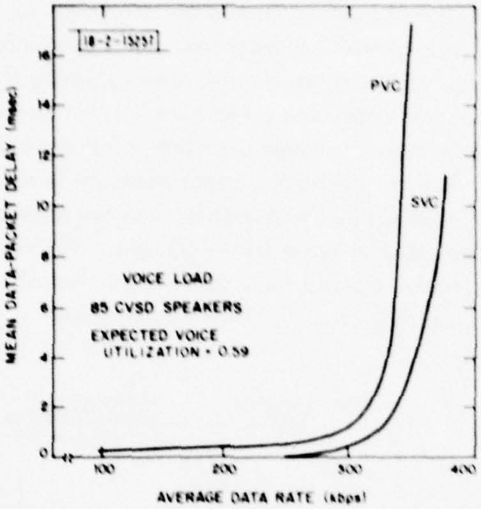


Fig. II-13. Comparison of mean packet delays from SVC and PVC simulations.

resulted in a mean data-packet delay of only 11 msec. The mean packet delay when the data arrival rate caused an 85-percent utilization was 29 msec. Total voice and data utilization for these two arrival rates was 91.5 and 93.7 percent, respectively. Mean data delay as a function of data arrival rate in both the PVC and SVC simulations is shown in Fig. II-13. The differences between SVC and PVC were rather small, and can be attributed to differences in the details of the two simulations. In both situations, the same fundamental characteristics of voice and data traffic were being exploited, so the fact that performance was similar was to be expected.

III. DEMAND-ASSIGNMENT MULTIPLE ACCESS TECHNIQUES

A. INTRODUCTION

The overall purpose of the Demand-Assignment Multiple Access (DAMA) study has been to investigate various satellite DAMA schemes in order to assess their suitability for use in future integrated voice/data networks. A DAMA scheme of particular interest is the Priority-Oriented Demand Assignment (PODA) scheme which was designed in the Atlantic Packet Satellite Experiment (APSE) program and which will be implemented in the EISN DAMA processors. During FY 78, the DAMA study focused on the problem of efficiently multiplexing the traffic from a large number of relatively narrowband ground terminals onto a wideband satellite channel. This configuration was of interest as a model of future Defense Communications Systems (DCS) which may include hundreds of ground stations. Since the advantage of statistically multiplexing a large number of users is limited, it appears that multiplexing must be effected at the satellite where the statistical properties of the large, aggregate traffic stream are available.

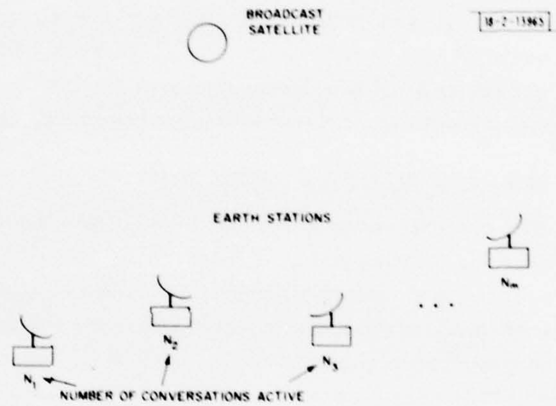


Fig. III-1. Configuration of satellite communications system.

The situation of interest is indicated in Fig. III-1, which depicts a satellite communications system with m ground stations and N_i voice callers in active conversation ("off-hook") at each ground station. The object is to achieve bandwidth savings by utilizing Speech Activity Detection (SAD) and allowing talkers to occupy satellite channel capacity only during talkspurts. Since the typical talker is silent more than 50 percent of the time during a conversation, the potential saving is substantial. However, experience with the circuit-switched Time-Assigned Speech Interpolation (TASI) system indicates that the number of talkers multiplexed must be greater than 40 to approach the full potential bandwidth saving. In the situation of interest here, N_i is typically less than or equal to 10, while $\sum_{i=1}^{m} N_i$ is greater than 40.

A brief discussion of the classical TASI system is appropriate at this point. This system multiplexes M talkers onto $c < M$ circuits by means of voice-activated switching. When a talker initiates a talkspurt, he seizes a circuit (if one is available) and holds it until the talkspurt is complete. If no circuit is available at talkspurt initiation, speech is lost until one of the ongoing talkspurts ends and a circuit becomes available. The number of circuits c must be large enough to maintain a sufficiently low fractional speech loss. The "TASI advantage" is defined as M/c, the ratio of the number of talkers to the number of circuits, and is generally quoted on the basis of a fractional speech loss of 0.5 percent. Assuming, for example, that each talker is issuing talkspurts 48 percent of the time, the TASI advantage is 1.25 for M = 10 and 1.67 for M = 40.

A promising three-part approach to obtaining efficient statistical multiplexing of speech in the situation depicted in Fig. III-1 has been identified and studied. The approach presupposes a DAMA scheme (such as PODA) which allows dynamically variable stream reservations. The three components of the approach are:

(1) Prediction of the number of active talkers at each station ahead by the time required to change the size of a stream reservation;

(2) Buffering of speech at each station, and trading of delay for TASI advantage;

(3) Provision of a backup mode for overflow traffic via a nonreserved channel excess.

A speaker activity prediction scheme has been derived and tested, and is described in Sec. B below. In trying to evaluate the effects of this prediction technique within a DAMA scheme, it was realized that asynchronous multiplexing and buffering of speech allows a trade-off which was not available in the circuit-switched TASI system. This issue is discussed in Sec. C. The third component listed above has not yet been examined in any detail.

B. SPEAKER ACTIVITY PREDICTION

The considerations above led to the conclusion that any DAMA scheme that successfully achieves the TASI advantage offered by the speech sources will have to incorporate some sort of reservation mechanism (such as the variable stream reservation capability in PODA) in order to insure that speech packets do not suffer excessive delays. Stream reservations are well matched to speech transmission requirements because they take cognizance of the non-bursty nature of the speech source. On the other hand, to make or change such reservations takes time (perhaps one or two round-trip times), which renders them suitable for new calls but not immediately useful for obtaining the speech source TASI advantage. In this connection, it seems obvious that the ability to make reservations without incurring large delays is critically dependent on an ability to predict future speaker activity by the amount of time it takes to change a reservation. Accordingly, a theoretical study of the predictability of speaker activity was undertaken.

The starting point for this study was the speaker activity model introduced by Weinstein.[5] This is a Markov model whose state transition diagram is depicted in Fig. III-2 where the average talkspurt and silence durations are denoted μ^{-1} and λ^{-1}, respectively. The quantity M denotes the number of active speakers. It is very well known that the optimum, least-squares prediction of the number of speakers at time t, given the past history of the number of active

Fig. III-2. Speaker activity model.

speakers in the interval ($-\infty < t' \leq s < t$), is given by the conditional expectation $E[n(t)|n(t'), -\infty < t' \leq s]$ where $n(x)$ denotes the number of active speakers at time x. In addition, the fact that $n(t)$ is a Markov process implies that this conditional expectation is the same as $E[n(t)|n(s)]$, i.e., all knowledge of the past of $n(t)$ prior to time s is summarized in the current value $n(s)$.

An explicit expression for this conditional expectation, as well as the mean-squared error of the optimum predictor, can be obtained. The starting point for this calculation is a partial-differential equation that the moment-generating function of $p_{j,k}(s,t)$ — the probability that $n(t) = k$ given that $n(s) = j$ — must satisfy. This moment-generating function is defined by

$$\psi_{j,s}(z,t) \equiv \sum_{k=0}^{\infty} z^k p_{j,k}(s,t) \tag{III-1}$$

and the partial-differential equation[6] is

$$\frac{\partial}{\partial t} \psi_{j,s}(z,t) = M\lambda(z-1)\, \psi_{j,s}(z,t) - (z-1)(\lambda z + \mu) \frac{\partial}{\partial z} \psi_{j,s}(z,t) \tag{III-2}$$

with the boundary condition

$$\psi_{j,s}(z,s) = z^j \quad . \tag{III-3}$$

Equation (III-2) subject to Eq. (III-3) can be solved by a technique due to Lagrange,[7] with the result

$$\psi_{j,s}(z,t) = \left\{ z \frac{1 + \frac{\mu}{\lambda} e^{-(\lambda+\mu)\tau}}{1 + \frac{\mu}{\lambda}} + \frac{\frac{\mu}{\lambda}[1 - e^{-(\lambda+\mu)\tau}]}{1 + \frac{\mu}{\lambda}} \right\}^j$$

$$* \left\{ z \frac{1 - e^{-(\lambda+\mu)\tau}}{1 + \frac{\mu}{\lambda}} + \frac{\frac{\mu}{\lambda} + e^{-(\lambda+\mu)\tau}}{1 + \frac{\mu}{\lambda}} \right\}^{M-j} \tag{III-4}$$

where $\tau = t - s$ denotes the prediction time.

The first term in braces in Eq. (III-4) is recognized as the moment-generating function of a binomial distribution with parameters $n_1 = j$ and $p_1 = [1 + (\mu/\lambda)\, e^{-(\lambda+\mu)\tau}]/1 + (\mu/\lambda)$. Similarly, the second term in braces is the moment-generating function of a binomial distribution with parameters $n_2 = M - j$ and $p_2 = [1 - e^{-(\lambda+\mu)\tau}]/1 + (\mu/\lambda)$. It now follows that $p_{j,k}(s,t)$ is the distribution of the sum of two independent binomial random variables with parameters (n_1, p_1) and (n_2, p_2). Finally, it is seen that the mean of the conditional distribution is the sum of the mean of its constituent binomial distributions, i.e.,

$$\mu_j(\tau) \equiv E[n(t)|n(s) = j] = n_1 p_1 + n_2 p_2$$

$$= \frac{M}{1 + \frac{\mu}{\lambda}} \left\{1 - [1 - (1 + \frac{\mu}{\lambda}) \frac{j}{M}]\, e^{-(\lambda+\mu)\tau}\right\} \quad . \tag{III-5}$$

Similarly, the independence of the constituent binomial distributions yields the mean-squared error of the optimum predictor as the sum of the variances of the constituent binomial distributions,

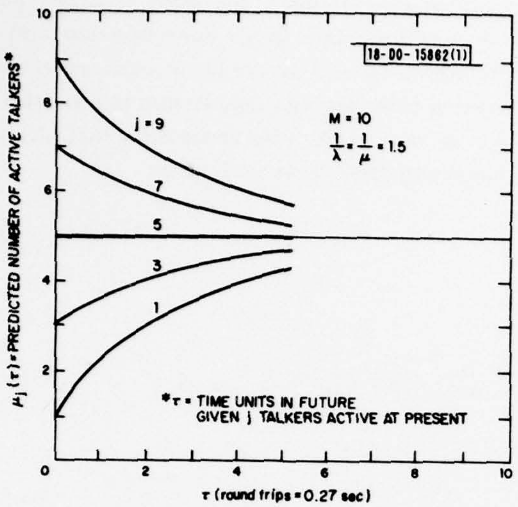


Fig. III-3. Optimum talkspurt predictor.

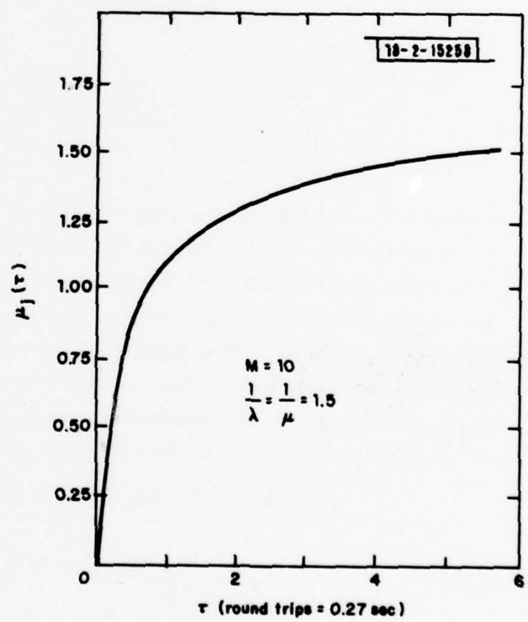


Fig. III-4. Prediction error of optimum talkspurt predictor.

$$\sigma_j^2(\tau) = E\{[n(t) - \mu_j(\tau)]^2 \,|\, n(s) = j\}$$

$$= n_1 p_1 (1 - p_1) + n_2 p_2 (1 - p_2)$$

$$= \frac{M \frac{\mu}{\lambda}}{(1 + \frac{\mu}{\lambda})^2} [1 - e^{-(\lambda+\mu)\tau}] \; \{1 + [\frac{j}{M}(\frac{\mu}{\lambda} - \frac{\lambda}{\mu}) + \frac{\lambda}{\mu}] \, e^{-(\lambda+\mu)\tau}\} \quad . \qquad \text{(III-6)}$$

Plots of Eqs. (III-5) and (III-6) for the case $\mu = 10$, $\mu^{-1} = \lambda^{-1} = 1.5$ sec are shown in Figs. III-3 and III-4. Note that when $\lambda = \mu$, $\sigma_j^2(\tau)$ is independent of j. Inspection of these curves indicates that reasonably good prediction (±1 speaker rms error) can be realized for prediction times on the order of a round-trip delay. This is very encouraging because it seems unlikely that, if this were not the case, the available TASI advantage could be realized without large delays.

A computer program simulating the talkspurt-silence behavior of any number of independent speakers has been written for the purpose of checking the validity of the speaker activity prediction technique reported above. The program does not use exponentially distributed talkspurt and silence distributions as was assumed for the theoretical results. Instead, talkspurt and silence distributions measured by Brady[8] of Bell Laboratories were used in order to check how well the real world matches the theoretical model. The simulation was used to generate the optimum least-squares predictor of future speaker activity, the rms error of the predictor, and the probability distribution of the number of active speakers.

The results of such a run for a prediction time of one round trip (0.27 sec) are shown in Table III-1 where: j denotes the number of active speakers at the current time; mu denotes the average number of speakers 0.27 sec in the future, given the current number of active speakers (this is the least-squares optimum prediction); sigma denotes the standard deviation of the prediction error; and p(j) denotes the steady-state probability of these j active speakers. The same parameters as derived from the the theoretical model of speaker activity are shown in Table III-2. An average talkspurt duration of 1.22 sec and an average silence duration of 1.33 sec (so that the fraction of time an individual talker is in talkspurt is approximately 48 percent) were used in deriving these numbers. It is seen that the agreement of theory and experiment is quite good.

C. TRADE-OFF BETWEEN TASI ADVANTAGE AND DELAY

The next logical step in this investigation of speech-oriented DAMA schemes would seem to be an investigation of how well a predictor would perform as part of such a scheme. To this end, a simulation of a simplified DAMA scheme employing prediction was written. In trying to evaluate the initial simulation results, it was realized that a fundamental trade-off exists between TASI advantage and delay. This trade-off is not dependent on any particular DAMA or prediction scheme, but seems to be a fundamental property of TASI schemes in which delay is admissible. This was not the case in conventional, circuit-switched analog TASI schemes, which probably accounts for the fact that this trade-off seems not to have been remarked upon previously in the literature.

An experimental investigation of the delay-vs-TASI-advantage phenomenon was undertaken using the setup depicted in Fig. III-5. A group of M speakers each generates talkspurts and

TABLE III-1

MEASURED TALKSPURT DATA

j	mu	sigma	p(j)
0	1.5209	1.2310	0.0015
1	2.2616	1.1734	0.0122
2	2.9596	1.1743	0.0548
3	3.6141	1.1632	0.1348
4	4.2553	1.1741	0.2180
5	4.9385	1.1773	0.2471
6	5.6153	1.1686	0.1894
7	6.2570	1.1720	0.1000
8	6.9429	1.2194	0.0330
9	7.7161	1.1839	0.0080
10	8.3055	1.2406	0.0008

TABLE III-2

THEORETICAL TALKSPURT DATA

j	mu	sigma	p(j)
0	1.6494	1.1736	0.0015
1	2.3044	1.1778	0.0137
2	2.9595	1.1820	0.0565
3	3.6145	1.1861	0.1382
4	4.2695	1.1903	0.2217
5	4.9245	1.1944	0.2437
6	5.5795	1.1986	0.1861
7	6.2345	1.2027	0.0974
8	6.8895	1.2068	0.0334
9	7.5446	1.2109	0.0068
10	8.1996	1.2150	0.0006

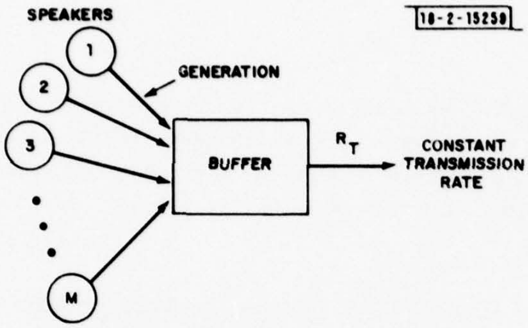


Fig. III-5. TASI advantage/delay measurement setup.

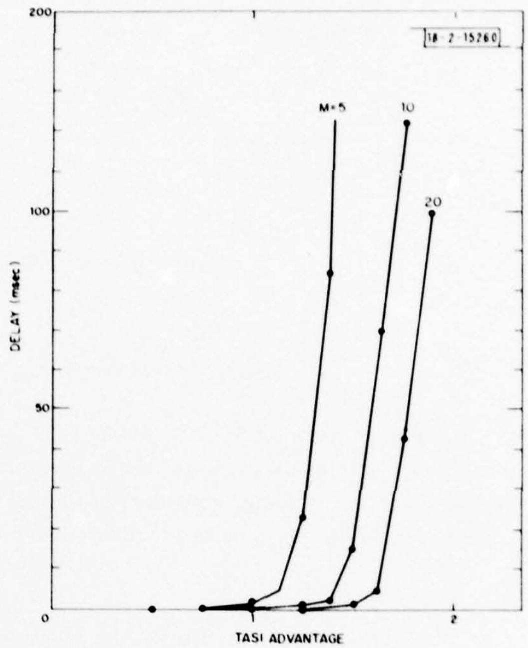


Fig. III-6. Delay as a function of TASI advantage for various numbers of off-hook callers.

silences according to the Brady statistics. The talkspurts are digitized and queued in a buffer to await transmission. No digits are produced during silences. The digits are removed from the queue at a fixed rate. The quantity to be measured is the average delay of digitized speech in the queue as a function of the total input transmission rate.

The results of simulating the above-described situation are summarized in the family of plots of average delay vs TASI advantage shown in Fig. III-6. TASI advantage is defined as $M/(R_T/R_G)$ where M denotes the number of speakers, R_T the transmission rate, and R_G the rate at which each speaker generates data when in talkspurt. Note that the denominator can be interpreted as the number of individual transmission circuits provided by the system, so that this definition is consistent with the conventional definition of TASI advantage given above. These curves clearly illustrate the fact that the TASI advantage available is a function of the delay that can be tolerated. It is interesting to note that the points where the curves break away from the zero-delay axis and begin to rise agree closely with the zero-delay TASI advantage points published by Bullington and Fraser.[9] For example, the latter claim a TASI advantage of 1.25 for 10 speakers and this is almost exactly where the breakaway point of the $M = 10$ curve of Fig. III-6 is located.

Another view of the trade-off between delay and TASI advantage, which places in evidence the comparison between asynchronous multiplexing (e.g., packet techniques) and traditional circuit approaches, is shown in Fig. III-7. Here, TASI advantage is plotted as a function of the number of off-hook callers, with delay as a parameter. The "no delay" case corresponds to the traditional circuit-switched TASI system. The improvement in TASI advantage which can

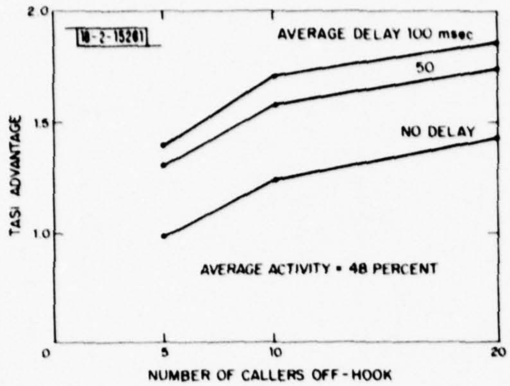


Fig. III-7. Effect of buffering on TASI advantage.

be attained by allowing 50 and 100 msec of average delay is apparent. For these curves, the percentage of time an average talker was active was assumed to be 48 percent. Note that the fractional activity during real conversations on a TASI system has been observed[9] to be somewhat less than 48 percent, so that the savings indicated are conservative.

D. SUMMARY

The current status of the DAMA study is summarized by the optimum prediction results and the TASI advantage vs delay curves discussed above. Current efforts are directed toward combining these techniques within a stream-reservation-oriented DAMA algorithm. The need for and effectiveness of setting aside a contention channel for overflow speech will be evaluated.

It should be emphasized that the results developed here with regard to prediction and the effect of delay on TASI advantage are quite general and not restricted to satellite systems. In particular, the TASI advantage vs delay trade-off summarized in Fig. III-7 leads to a very provocative idea with regard to any terrestrial or satellite system which allows asynchronous multiplexing and buffering (packetized or via some other technique) of speech. The curves indicate the potential for achieving a full network-wide TASI advantage even when only small concentrations of users can be multiplexed at individual network locations.

IV. PRELIMINARY EXPERIMENT PLAN† FOR THE EXPERIMENTAL INTEGRATED SWITCHED NETWORK

A. INTRODUCTION

1. Purpose and Background for the Experimental Wideband Network

A wideband Experimental Integrated Switched Network (EISN) is currently being developed under the joint sponsorship of the Defense Communications Agency (DCA) and the Defense Advanced Research Projects Agency (DARPA). Organizations currently participating in the program include: Bolt Beranek and Newman (BBN), Communications Satellite Corporation (COMSAT), Information Sciences Institute (ISI), Linkabit Corporation, M.I.T. Lincoln Laboratory, and SRI International. Several additional organizations will join the program as contractors are selected for various subsystems now under competitive bid. The network will provide a unique experimental capability for the investigation of systems issues involved in a communications facility which includes wideband satellite and terrestrial links and which carries large volumes of voice and data traffic. Areas for experimental investigation include: (a) demand-assignment strategies for efficient broadcast satellite communications; (b) packet speech communication in a wideband, multi-user environment; (c) alternate integrated switching techniques for voice and data, with particular attention to the advantages and disadvantages of packetized voice communication relative to more-conventional circuit techniques; (d) rate-adaptive communication techniques to cope with varying network conditions; (e) routing of voice and data traffic; (f) digital voice conferencing; and (g) internetting between satellite and terrestrial subnetworks. Some of these areas are of more-immediate concern to one or the other of the two sponsoring agencies. However, the problem areas are so interrelated and complementary that all are of interest to both agencies.

Recent efforts related to the current wideband network program include:

(a) The DARPA Packet Speech Program in which the ARPANET was employed for development and demonstration of packet speech communication techniques;

(b) DCA-sponsored development of hybrid (packet and circuit) switching techniques for voice and data;

(c) The Atlantic Packet Satellite Experiment (APSE), jointly sponsored by DARPA and DCA, along with the British Post Office and the Norwegian Telecommunications Authority, which included the development of the Priority-Oriented Demand Access (PODA) class of satellite demand-assignment algorithms.

Existing networks such as the ARPANET and the Atlantic Packet Satellite Net do not have sufficient capacity to permit experiments on a scale large enough to realistically represent the multiple and varied user environment that will be typical of future military communications networks.

† A more-detailed version of this plan has been submitted as a separate document.

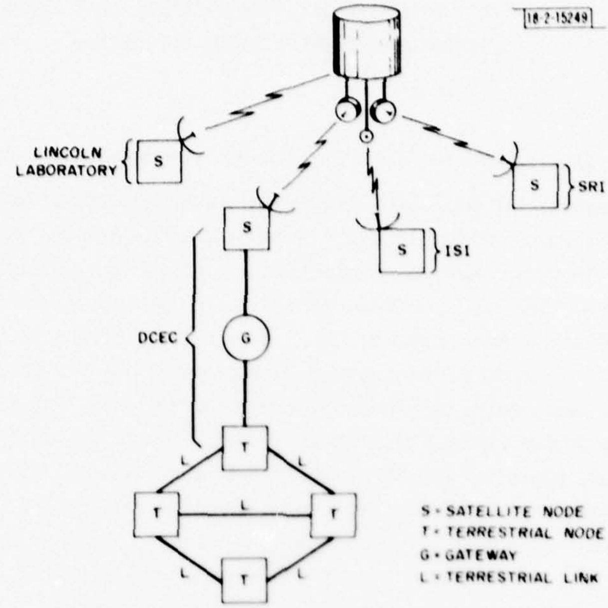


Fig. IV-1. Topology for experimental wideband network.

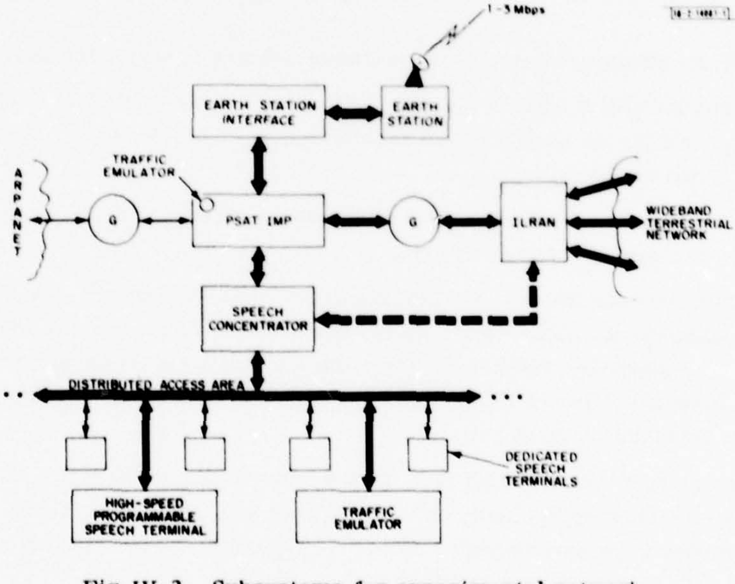


Fig. IV-2. Subsystems for experimental network.

The wideband integrated voice/data network is intended to provide a more-realistic environment for experimental investigation and demonstration of the advanced communications techniques listed above. The complete test bed will include wideband satellite and terrestrial links and switching nodes; access facilities, including concentrators and terminals; traffic emulation modules; various monitoring and support subsystems; and gateways for satellite/terrestrial internetting experiments. The implementation of major network subsystems will proceed over the next few years. The intent is to evolve the network in such a way that experiments can begin as soon as the first major subsystem is operational.

2. Summary of Preliminary Experiment Plan

A stand-alone document representing a preliminary effort at an overall experiment plan for the wideband network has been delivered to DCA and DARPA under separate cover. The experiment plan is expected to evolve and become more detailed as the network evolves over the next few years, and the plan will be revised and expanded accordingly. This section represents a somewhat abridged version of the planning document. The intention is to include all major areas which are covered in the planning document, but to omit some of the details of schedules, equipment requirements, and issues which require coordination among the program participants. These details are of less general interest and will be subject to changes as the plan is coordinated among the sponsors and contractors.

An important purpose of the preliminary planning document is to describe a development plan for the network test bed. A description of this plan, including a discussion of the functions of the various subsystems, is given in Sec. B below. The other primary purpose of the planning document is to define and categorize systems experiments. The earliest experiments, which will be directed at test and validation of basic system functions and capabilities, are discussed briefly in Sec. C. Finally, advanced systems experiments are the subject of Sec. D. A broad set of experimental areas is defined, and objectives and performance measures are described for each class of experiments. Scheduling for test-bed development and for systems experiments is touched on only briefly here. More-detailed scheduling information is presented in the separate planning document.

An important role of the planning document is to stimulate further interaction and coordination among the program participants. To this end, a set of unresolved issues with respect to test-bed development, system validation, and advanced systems experiments are listed in the planning document. Discussion of these issues has generally not been included here.

B. DEVELOPMENT PLAN FOR THE EXPERIMENTAL WIDEBAND NETWORK

The experimental network will include a wideband satellite network, a wideband terrestrial network, access facilities including concentrators and terminals, and internet gateways. The satellite net will include four earth stations with planned locations at: Defense Communications Engineering Center (DCEC), Reston, Virginia; ISI, Marina del Rey, California; Lincoln Laboratory, Lexington, Massachusetts; and SRI International, Palo Alto, California. At least one of the terrestrial switching nodes will be collocated with a satellite station, but the locations of all terrestrial nodes have not yet been specified. A topology for the satellite and terrestrial nodes is shown in Fig. IV-1. For illustration purposes, the DCEC location is depicted as the site of both a satellite and a terrestrial node, as well as a gateway interconnecting the two. Locations of the other three terrestrial nodes are unspecified. Figure IV-2 shows a projected configuration for subsystems to be available at a network location which is the site of both satellite and terrestrial nodes. The functions of each subsystem are discussed below.

1. PLURIBUS Satellite Interface Message Processor (PSAT IMP)

The satellite network communication protocols, including the Demand-Assignment Multiple Access (DAMA) protocols for the broadcast channel, will be implemented in the PSAT IMP which is a flexible device that can be programmed to serve a variety of experimental needs. Hardware and software development of the PSAT IMPs is the responsibility of BBN. The satellite IMPs developed by BBN for the APSE carried out similar functions, but were implemented as single-processor machines. The single-processor implementation was sufficient to support the 64-kbps broadcast channel employed in the APSE. The satellite channel in the current experiment will be designed to accommodate approximately 3 Mbps, and a multiprocessor approach referred to as the PLURIBUS technology will be employed to satisfy this significantly higher throughput requirement.

The demand-assignment algorithms to be implemented in the PSAT IMPs are Time-Division Multiple Access PODA (TPODA) and Contention PODA (CPODA). In TPODA the reservation subframe is shared by fixed-slot allocation, while CPODA uses slotted-ALOHA type contention in the reservation subframe. The PODA algorithms provide a flexible facility for supporting a variety of DAMA-type experiments. For example, the stream capability with fixed reservations can be utilized to emulate a fixed-TDMA scheme for basic satellite channel tests. As in APSE, the PSAT IMPs will have internal capabilities for network experiments and measurements, consisting of fake hosts which can be activated to serve as artificial traffic sources and sinks, and other fake hosts which can be activated to collect and transmit cumulative statistics on specified performance parameters. As was the case in the SIMP-3 version of the Atlantic Satellite Net, the PSAT network will be implemented as a stand-alone structure, capable of functioning independent of the ARPANET. However, facilities existing in host computers on the ARPANET will be utilized for experimental purposes as appropriate.

2. Earth Station Interface (ESI)

The ESI provides an interface between the PSAT IMP and a 70-MHz intermediate frequency (IF) line into the satellite earth station. Linkabit Corporation has responsibility for ESI equipment development. ESI modules include a packet/burst controller which accepts packets from the PSAT IMP and forms bursts of digital data for transmission, a burst modem, and a command/monitoring unit. The ESI will be required to operate at rates from 386 kbps to 3.088 Mbps. A similar interface developed for APSE operates in the range 16 to 64 kbps, so that a significantly scaled-up capability is required. The ESI will allow a multi-rate transmission mode within a burst. Modulation modes of Binary and Quaternary Phase Shift Keying (BPSK and QPSK) will be provided, as will optional error-correcting coding modes.

3. Earth Station and Channel

The earth station and channel will be purchased from a carrier on the basis of competitive bids. Earth station equipment will interface with the ESI at 70 MHz IF, and will include low-noise receiving amplifier, high-power transmitting amplifier, and antenna. A leased channel on a domestic satellite is to be provided. To minimize interference and siting problems, operation in either the 12- to 14-GHz or 18- to 30-GHz frequency bands is preferred. However, because of the current lack of satellites operating in these bands, it may be appropriate to provide initial capability at 4 to 6 GHz in conjunction with a phased upgrading to the higher frequencies. The earth station and channel will be required to provide sufficient signal-to-noise ratio to support bit rates up to 3.088 MHz at specified error rates.

4. Integrated Local/Regional Access Node (ILRAN)

The ILRANs are the switching/multiplexing nodes for the wideband terrestrial subsystem. A contractor is currently being selected by the DCA to study advanced integrated systems concepts including hybrid and packet switching techniques, and to design a terrestrial network (including ILRANs) to allow experimental evaluation of these concepts. Assuming successful completion of the design study, four ILRANs will be constructed. The ILRANs will be designed to be interfaced with access-area user terminals, hosts, gateways, and concentrators. In addition, each ILRAN will have an associated module for generation of emulated traffic within the terrestrial network.

5. Access Facilities

Access facilities for the wideband network will include terminals, access areas, host computers, speech concentrators, and traffic emulators. The provision of facilities for multi-user voice access is an essential item in the wideband experiment, and will receive much attention over the first few years of the program. Much of the need for data traffic can be filled by traffic emulation, and access for real data traffic can be provided by standard host computers, as in the ARPANET. Voice experiments require subjective evaluation, and thus are more dependent on access for real voice users. In addition, the design of voice terminals, access areas, and speech concentrators is, in itself, an important topic for investigation in the wideband program. A heavy emphasis on voice problems is thus expected during the early phases of access facilities development. The associated issue of whether access for data traffic should be handled separately or integrated with voice access traffic remains open at this time.

a. Access Area

The access area provides the physical medium for collection and distribution of terminal data. A topology and a set of protocols must be specified in the access-area design. Lincoln Laboratory is currently studying (under DARPA sponsorship) access-area architectures, with particular attention to distributed approaches. During FY 79, Lincoln will design and validate a pilot access area capable of supporting a few real voice terminals as well as a substantial volume of emulated traffic. To allow early speech experiments on the wideband network, this pilot access area will be interfaced to the wideband network through a minicomputer host. This pilot access area is expected to be the forerunner of a wideband access area capable of handling 50 to 100 terminals.

b. Speech Concentrator

The purpose of the speech concentrator is to collect the digital voice outputs from individual voice terminals and form them into one wideband data stream which is presented to a network node. It also performs the inverse process of splitting the return stream for transmission to the terminals. The concentrator has the task of transforming the local protocols of the voice terminals into the line transmission or packet protocols required by the wideband network node. It may also perform functions such as silence detection and rate control. Initially, the concentrator will be connected to the PSAT IMP, with a software gateway in the concentrator performing the necessary protocol conversion. Later, as shown by the dotted line in Fig. IV-2, a concentrator/ILRAN connection will also be implemented. It should be noted that the ILRAN switch/multiplexer will be designed to include its own concentration function, and will be able to accept inputs from individual terminals as well as from concentrators. Voice experiments in the terrestrial network therefore might not require a speech concentrator. However, a concentrator/ILRAN connection will allow the access area and concentration facilities developed primarily for the satellite network to be utilized effectively in terrestrial network-based experiments.

The requirements for speech concentration have been the subject of a study during FY 78 at Lincoln Laboratory. The study has addressed specifically the separation of functions among concentrator, access area, and terminals. No specific schedule has been set for development of a speech concentrator capable of handling hundreds of terminals, as will eventually be required in the wideband network. However, early speech experiments will utilize only a few voice terminals, and the concentrator function for these experiments will be handled in a standard minicomputer, such as a PDP-11. This limited-capacity concentrator will be referred to here as the "miniconcentrator."

c. Speech Terminals

Speech communication experiments will form an essential part of the wideband network program, and speech terminals of various types will be utilized as the program proceeds. Speech terminals include the speech algorithm processor or vocoder function, as well as an access-area interface and suitable dial-up and ringing functions. Speech terminal development is not viewed as a primary focus of the wideband network program. However, reference is made here to current efforts sponsored by DARPA under the Packet Speech Program which will provide support for the wideband experiment in the speech terminal area.

Figure IV-2 shows both dedicated and programmable terminals in the access area. Dedicated terminals will run fixed vocoder algorithms and interact with the network in a fixed way. Currently available voice processors for dedicated operation include wideband encoders such as Continuously Variable Slope Deltamodulator (CVSD) devices, and a more-limited number of narrowband processors such as the Linear Predictive Coding (LPC) devices used in APSE. A current DARPA-sponsored program at Lincoln Laboratory and Texas Instruments (TI) is directed at developing a small, cheap, narrowband voice processor which can be deployed in large numbers in the wideband experiment. A prototype of such a unit, which is based on Charge-Coupled Device (CCD) technology and implements a channel vocoder algorithm, will become operational during FY 79.

A limited number of high-speed programmable terminals will be used for working with newly emerging voice algorithms and experimental techniques for terminal/network interaction. Of particular interest here are terminals which can communicate at varying bit rates, depending on network load. Voice algorithms of the embedded coding type are of special interest in this regard because of their ability to rapidly change rates. Such algorithms, which are currently under development at Lincoln and elsewhere, will be tested on high-speed programmable devices. Currently available programmable processors such as the Lincoln Digital Signal Processor (LDSP) will be able to support such experiments in a limited way. However, the complexity of some of the proposed new algorithms (e.g., embedded coding) is such that more-powerful programmable processors will be needed in the long run.

A study of the voice-terminal/access-area interface issue is currently being initiated at Lincoln. The design and construction of interfaces to support experimental work in this area will be carried out as part of this effort.

d. Access-Area Traffic Emulator

The function of the access-area traffic emulator is to simulate the traffic loading effect of many voice terminals in the access area. Since voice traffic flow control is envisioned as being implemented either in the speech concentrator or at the voice terminal itself, traffic emulators resident in the PSAT IMP cannot satisfy this requirement. Rather, the emulator must act via the access area so that the concentrator's flow-control mechanisms are exercised. The required access-area traffic emulation is expected to be effected in software either in conventional PDP-11 type machines or in currently available high-speed processors. Lincoln has the responsibility for implementing a software capability for access-area voice traffic emulation. The emulation will include call initiation/termination and the use of speech activity detection for each of the simulated speakers. The emulation software will include voice flow-control techniques such as bit-rate selection at dial-up, dynamic modification of vocoder rates during conversations, and the embedded coding technique.

6. Gateways

Gateways allow internet communication between the wideband network and other networks. Two gateways are shown in Fig. IV-2, although other internet connections will probably be implemented during the course of the experimental program. An example of particular interest is a gateway to the ARPA Packet Radio Net.

An initial capability for internet communication between the PSAT IMP and the ARPANET will be implemented by BBN in the form of a software module in the PSAT IMP processor. This internal PSAT "mini-gateway" will support transfers of experimental data and control between PSAT IMP fake hosts and TENEX-based facilities residing on the ARPANET. However, experiments involving real internet (data or voice) traffic traversing the ARPANET and wideband satellite net will require a full gateway capability similar to the PDP-11 configuration used in the APSE.

The PSAT IMP/ILRAN gateway will be used for wideband terrestrial/satellite internetting experiments. Design of this wideband gateway should begin after the initial ILRAN capability has been established.

7. Wideband Terrestrial Links

When the ILRANs are eventually delivered to network locations, wideband terrestrial links will be needed to interconnect them. The earliest need for such links is projected to occur during FY 82.

8. System Control and Monitoring

System-control and monitoring support facilities will be needed throughout the experimental program. The monitoring and control technology developed for APSE will be carried over by

BBN for use with the PSAT network. A monitoring and control fake host will be implemented in the PSAT IMP processor, and communication capability between this fake host and TENEX monitoring and control programs will be established. The ILRAN contractor is required to develop a Network Monitoring Center (NMC) as part of the terrestrial network implementation. Provision must be made for the control and monitoring of satellite/terrestrial internet experiments. Options include physically combining the network control centers or establishing a communications path between them.

9. Subsystem Development and Installation Schedule

A tentative schedule for development and installation of the wideband experimental facility is presented in some detail in the separate experiment planning document. A key milestone date on that schedule – 1 January 1980 – is worth calling attention to here.

Basic satellite subsystems – including PSAT IMPs, ESIs, and earth stations – will have been installed at the first two network locations (probably ISI and Lincoln) by 1 January 1980. In addition, an access facility including a prototype access area, one or two narrowband voice terminals, a traffic emulation capability, and a miniconcentrator will be available at one of the sites (Lincoln). This date thus marks the starting point for experiments with the wideband satellite subnetwork, including emulated data and voice traffic as well as live voice.

The terrestrial network switches (ILRANs) will be developed in a parallel program which extends through FY 81. Construction and initial testing of the ILRANs is due to be completed at the end of FY 80, and systems experiments with the four ILRANs interconnected within the contractor's test facility will proceed during FY 81. Delivery of the ILRANs to specified network locations will occur early in FY 82. This represents another key milestone in the network development schedule.

C. SYSTEM VALIDATION

1. Introduction

Experiments for the wideband network can be roughly divided into two classes: (a) system validation experiments, which verify or measure the performance of a network component or function; and (b) system concept experiments directed at exploring some specific issue or problem in integrated voice/data networking. The concept experiments motivate the building of a test-bed system and are designed to provide answers to important systems questions. The validation experiments have as a goal the performance verification of the test-bed facility on which the concept experiments will be carried out.

2. Validation Experiments

The preparation and execution of validation test plans are generally the responsibility of the subsystem contractors. However, the separate experiment plan document gives a summary review of the minimal objectives that should be satisfied by these validation plans. That review includes discussion of validation experiments relevant to the satellite subsystem, the terrestrial subsystem, access facilities, and gateways. The review is not given here, since many of the items require specific coordination with individual contractors.

D. ADVANCED SYSTEMS EXPERIMENTS

Advanced systems experiments are defined and discussed here. Seven classes of experiments are considered in Secs. 1 through 7 which follow; these are:

(1) Demand-Assignment Multiple Access (DAMA) Strategies

(2) Multi-user Packet Speech Communications

(3) Advanced Switching/Multiplexing Techniques for Voice/Data Integration

(4) Rate-Adaptive Communications Techniques

(5) Routing

(6) Conferencing

(7) Internettting

It should be recognized that there is significant overlap among the experiment areas. This overlap can be used to advantage in that particular experiments will provide results in more than one area. For each class of experiments, the following items are discussed:

(1) Introduction and background,

(2) Experiment objectives,

(3) Experimental approach and requirements, and

(4) Performance measures.

The separate experiment plan document includes more-detailed discussion of each experiment area, and presents a tentative schedule for each class of experiments.

1. Demand-Assignment Multiple Access (DAMA) Strategies

a. Introduction

Broadcast communication satellites have a unique potential to efficiently support general-purpose communications, including both data and voice, among a large and diverse set of users. Cost-analysis studies have shown that substantial savings can be achieved with systems employing hundreds of earth stations. However, a prerequisite for the achievement of these savings is the development of flexible DAMA techniques that allow one to exploit the broadcast nature of the satellite channel. The APSE program has provided an impetus for the development of such DAMA techniques and a test bed for their evaluation. Techniques investigated in APSE include fixed-TDMA, round-robin TDMA, slotted-ALOHA, and several variations of PODA. A basic requirement of these algorithms is the ability of the satellite earth station equipment to transmit and receive in a flexible burst-TDMA mode. The demonstration of this capability has been an important achievement of the APSE program. Of the DAMA schemes tested, the PODA algorithms appear to be the most promising in terms of their ability to handle the expected mix of user requirements, including block (data) and stream (voice) traffic. However, the PODA algorithms also place the most-stringent requirements on the DAMA processors in terms of processing speed and memory.

DAMA-related experiments in the APSE have been limited in scope because of the restricted (64-kbps) bandwidth of the satellite channel. This restriction has prohibited full testing of the ability to handle mixes of voice and data traffic, and of the reservation-handling capability of the DAMA processor. The wideband satellite network will provide an environment for DAMA experiments with traffic loads more representative of expected user communities. The PODA implementation in the PSAT IMPs will provide an effective and flexible facility for experimenting with realistic traffic volumes and mixes, as well as with various DAMA schemes.

b. Experiment Objectives

DAMA experiments will be aimed at testing and evaluating the effectiveness of various multiple-access protocols in environments typical of the satellite portions of future military communications systems. Experiments will be directed at the investigation of techniques for (1) efficiently sharing the satellite channel capacity among many earth stations, and (2) taking advantage of the differing statistics and transmission requirements of voice and data traffic to achieve efficient statistical multiplexing. For example, there is strong interest in investigating and developing schemes for achieving the TASI advantage by statistical multiplexing at the satellite in cases where only a few voice users are present at each ground station. The feasibility of achieving this type of statistical multiplexing, which may require very rapid variation in satellite capacity assignments, will be investigated. The attendant advantages and costs of highly responsive DAMA schemes should be compared against simpler schemes which offer slower variation in channel allocation. Another objective is to compare the effectiveness and processing requirements of distributed and centralized control of the channel assignments in order to select an appropriate mix of these control schemes for future systems.

Of major concern in all cases are the earth station equipment requirements needed to support the algorithms (hardware size and cost, software complexity) and the robustness of the systems with respect to earth station error or failure, transmission channel errors, etc. An important problem area is that of verifying that selected DAMA schemes can actually function efficiently with large volumes of real traffic without developing lockup problems or other limiting effects.

c. Experimental Approach and Requirements

The PODA algorithm will be used as a flexible experimental tool for carrying out the DAMA experiments. Traffic emulation in the PSAT IMPs will serve as the primary means for generating the traffic mixes required to test DAMA performance and processing requirements under a variety of conditions. Real voice and data traffic will be combined with the emulated traffic as appropriate. The measurement fake-host facility in the PSAT IMP will be used for measuring the patterns of delays, lost packets, etc. for the various experiments. These data will be delivered to a TENEX host for analysis and display.

Two important sequences of experiments in the DAMA area are identified and discussed in some detail in the planning document. The first concerns techniques for the efficient handling of full-duplex speech users, in the case where many nodes are each supporting a small number of speakers. The second sequence is directed at assessing the effects of stressing the DAMA algorithm under different dynamic response constraints.

d. Performance Measures

Performance measures for voice include delay distributions, percentage packet loss, and the efficiency with which the burst nature of speech can be exploited. Data performance measures include average delay, buffer requirements, and probability of packet misdelivery. For the overall system, the efficiency of utilizing the satellite channel resource is of prime importance. The processing cost of achieving high channel utilization for various user mixes must be determined. The robustness of the system is an important performance measure. The system must be able to accommodate synchronization loss or hardware failure at individual stations without seriously compromising overall network performance. In addition, temporary data overloads must be accommodated without catastrophic system failure. Finally, the system should exhibit fairness in its ability to provide the same quality of service to all users of equal priority without allowing individual users to capture a disproportionate share of the channel resources.

2. Multi-user Packet Speech Communications

a. Introduction and Background

Packet speech communication has become a subject of intense interest and activity over the past several years. Factors motivating this interest include: (1) The development and demonstrated advantages of packet techniques for data communications; (2) the potential of packetized techniques for increasing channel utilization by exploiting the bursty nature of speech transmission; and (3) the possibility for the integration of voice and data in a common system based on packet switching. The basic feasibility of packet speech communication was first demonstrated in experiments on the ARPANET under the DARPA Packet Speech Program. Developments under this program included voice protocols for packet networks and techniques for maintaining speech communication quality in the face of the delay dispersion inherent in packet transmission. Later, packet speech communication techniques for a broadcast satellite environment were developed in the APSE.

A recent economic study conducted by Network Analysis Corporation (NAC) concluded that packet switching is potentially the most cost-effective technique for the integrated switching of voice and data. Support for this conclusion will require a demonstration of the feasibility of packet voice on a large scale. However, packet speech experiments to date have accommodated only a few voice users because of channel capacity limitations in the ARPANET and Atlantic Satellite Net.

b. Experiment Objectives

A primary objective of the experimental program is the development and demonstration of packet speech techniques for a wideband, multi-user environment. Included in this objective is the investigation of the effectiveness of packet techniques in achieving efficient statistical multiplexing of voice transmissions from a number of users. A prerequisite for multi-user packet speech experiments is the development of appropriate access facilities including concentrators, access areas, and packetized voice terminals. In fact, the investigation of advanced access and concentration techniques is in itself an important objective of the current experimental program. Distributed architectures for the access area, where terminals are interfaced to the concentrator through a common cable or bus, are of particular interest in this regard. In addition, the design of terminals and access facilities must reflect the need for eventual compatibility with privacy/security requirements. Experiments directed at demonstrating such compatibility should be included in the program.

c. Experimental Approach and Requirements

Packet speech experiments will be phased in a manner consistent with the development of the network test bed. The initial voice experiments will make use of two voice terminals in the same access area, interfaced to the PSAT IMP by means of a miniconcentrator. Loop-back tests through the PSAT IMP will be followed by tests where packets are transmitted up to the satellite and back to the same earth station. These tests will establish the effectiveness of the interfaces and protocols between the concentrator and the PSAT IMP, and will verify the dialing and signaling protocols which allow call setup through the satellite channel. Emulated traffic will then be combined with real voice traffic in order to determine the effects of loading the satellite channel near capacity. Fake hosts in the PSAT IMP will provide a source of emulated traffic characteristic of multiple-voice and/or data users.

When a second set of access facilities becomes available, these experiments will be repeated with two-way conversations between sites. Access-area traffic emulation will be used to test multiple call-handling capability and flow-control strategies in the concentrator, and to establish the throughput limitations of the complete path from access area to access area. A series of experiments will be run to test alternate voice multiplexing strategies in the concentrator. Schemes for packet aggregation, where speech packets from two or more terminals are combined into larger packets for transmission on the wideband net, will be developed and tested.

These experiments will later be scaled up to include a large number of real voice terminals interfaced to the network through a wideband concentrator. Although the discussion of packet speech experiments here has been focused on the satellite network, a similar set of experiments will be run later in the program in the terrestrial and internetted environments.

Experiments involving privacy/security will follow the initial developments of a packet speech capability. A requirement for such experiments is the availability of real or simulated packetized privacy devices. The BCR class of encryption control devices should be considered as a primary candidate for satisfying this requirement in the satellite channel and over wideband terrestrial trunks. Access-area privacy is an important topic for which a variety of options need to be considered.

d. Performance Measures

Performance measures for packet voice include delay distributions and percentage packet loss, and their effect on speech quality. The efficiency with which speech users can be statistically multiplexed is of essential importance. It must be determined to what degree the TASI advantage can be attained, and at what cost in packet overhead, switch processing requirements, and buffer storage. The call-handling capability and achievable throughput for the access area, concentrator, and satellite network need to be assessed individually to insure that a proper match is designed and resources are not wasted. Reliability and ease of use of the packet speech system will also be important concerns.

3. Advanced Switching/Multiplexing Techniques for Voice/Data Integration

a. Introduction and Background

The trend in military communications is toward all-digital networks which will serve large volumes of voice and data traffic. Problems of fundamental concern are (1) the economics of serving voice and data applications on a common integrated communications system, and (2) the comparison of alternate switching technologies for integrated voice and data networks.

Switching technologies to be considered include advanced packet and hybrid (combination of circuit and packet) approaches, as well as more-traditional circuit-switched methods. Analysis, simulation, and ARPANET experience indicate that packet techniques provide considerable advantages for the handling of interactive data traffic. The previously cited NAC study concluded that packet switching is potentially the most cost-effective technique for the integrated switching of heterogeneous traffic including voice, interactive data, and bulk data. Hybrid techniques were also shown to have significant advantages over all-circuit techniques. The key factor which separated packet and hybrid systems was the potential ability of packet systems to achieve approximately a 2:1 savings in bandwidth utilization [the so-called Time-Assigned Speech Interpolation (or TASI) advantage] for voice by avoiding transmission during silent intervals. The extent to which this savings can actually be achieved in a large, distributed network remains to be determined. Another issue to be investigated is the effectiveness of fast virtual-circuit switching techniques compared with packet approaches in achieving channel savings during speech pauses. Finally, interoperability between advanced switching techniques and existing network strategies, and the problems and costs of transition, are subjects of concern.

The efficient integration of voice and data is probably the most central issue in the wideband program, and pervades all the experiment areas. This section focuses mainly on the issues of voice/data integration in the terrestrial network or ILRAN system, since it is here that there are two specific switching technologies – hybrid and all-packet – which have to be examined.

b. Experiment Objectives

Recent analyses have indicated that advanced switching technologies, including packet and hybrid (combination of circuit and packet) approaches, have the potential for efficient integration of heterogeneous traffic types, with significant cost savings due to the sharing of switching and transmission facilities. However, the feasibility of such techniques has not yet been demonstrated on a large scale. A key purpose of the wideband program is to achieve such a demonstration, and to carry out an implementation exercise which develops the switching concept in sufficient detail to serve as a prototype for a future integrated military network. For all-packet systems, the development, demonstration, and cost analysis of packet voice communications on a large scale are subjects of key concern. For hybrid systems, the implementation and integration of packet and circuit switching into a common system must be demonstrated. These demonstrations will provide a sound basis for comparison between packet and hybrid techniques.

The objectives of this portion of the experiment are:

(1) Provide a flexible test bed for evaluating different techniques of voice/data integration both within the terrestrial network and in conjunction with the satellite network.

(2) Perform a series of experiments which will allow a quantitative evaluation to be made of the advantages and disadvantages of both techniques.

c. Experimental Approach and Requirements

The principal vehicle for examining the voice/data integration issues will be the ILRAN system consisting of four flexible switching nodes connected by wideband trunks.

The experiments will be conducted in two parts. The first part will consist of a 3-year development and test program to be conducted by the ILRAN contractor at his facility. In the second part, the equipment will be integrated with the satellite network. Experiments in this second area are described in Sec. 7, "Internetting."

The initial 3-year effort of the ILRAN contractor is, in turn, divided into three phases. In the first phase, the contractor will study the promising voice/data integration techniques including at least the packet and hybrid concepts. For each promising concept, the contractor will identify the key features which should be experimentally evaluated to assess their relative advantages and disadvantages. These experiment requirements will then be used as a guide for designing a flexible experimental switching network test bed to carry out the experiments. The contractor will also develop a comprehensive, general test plan describing the concept experiments to be performed on the terrestrial network.

In the second phase, the contractor will build and test the system designed in the first phase. He will also elaborate on the general concept test plan to develop detailed acceptance and concept test procedures. In the third phase, the detailed concept test plan will be carried out at the contractor's facility.

d. Performance Measures

Performance measures for voice include delay distributions and percentage packet loss and their effect on perception, as well as the number of users that can be handled for each scheme. The effectiveness of each scheme in exploiting speech-activity detection to save on channel utilization during pauses is of particular importance, since the packet-vs-circuit comparison hinges largely on this issue. For data traffic, delay distributions and buffer storage requirements as functions of voice traffic load and strategy for servicing the mixed traffic are of particular interest. In all cases, network throughput rates, channel utilization characteristics, and switch processing requirements will be critical performance measures.

4. Rate-Adaptive Communications Techniques

a. Introduction and Background

Conditions in an operating communications network are in a continual state of change. Traffic levels fluctuate as new users enter and leave the system, and as the capacity requirements of individual users vary with time. Link capacities may vary due to jamming or fading. Portions of the network may become temporarily unavailable due to equipment failures. An essential aspect of network operation is the ability to maintain the best possible service in the face of changing conditions. Data networks such as the ARPANET include routing and flow-control algorithms designed to cope with changing conditions. In an integrated voice/data network, an important additional opportunity for effective adaptation is presented because voice can be communicated at a variety of rates with different degrees of fidelity. Assuming the availability of a sufficiently flexible speech coder, voice bit rates could be adjusted up or down to match the capacity available in the network at a given time.

One approach to voice-rate control is to assign a rate to each user entering the system based on network loading at the time of dial-up and on the priority of the user. Ultimately, a user would be blocked if the network was unable to support him at his lowest bit rate. A second approach would provide more-rapid response to changes in network status by allowing terminals to change rates during conversation based on control signals from the network. A third approach permits essentially instantaneous reaction to network overloads by allowing some of the speech bits in transit to be discarded by network nodes along the communication path.

This third approach requires an "embedded coding" vocoder of the type originally proposed at the Naval Research Laboratory. In this technique, the voice signal is encoded into packets of different priorities. The lower-priority packets may be discarded at any time without affecting the intelligibility of the speech, although there will be some degradation in speech quality. In any vocoder frame, the speech synthesizer will use all the packets that arrive, and fill in for all those which are missing. This results in varying quality levels as the priority level of the arriving packets changes. To insure that intermediate nodes are not overloaded with packets that are later discarded, the embedded coding technique is combined with an end-to-end flow-control algorithm where the receiver notifies the transmitter of the current received rate, and the transmitter sends only those packets likely to be received at the destination.

During FY 78, an embedded coding vocoder based on a synthesis of channel vocoding and sub-band coding methods was developed at Lincoln Laboratory. This vocoder is capable of operating at rates from around 2 to 20 kbps with quality and robustness increasing with rate, and without perceptible transients due to rate switching. In addition, a study via simulation of the effects of end-to-end flow control in a multi-hop network with embedded coding of voice has been carried out. Control mechanisms which allow rapid adaptation and stable network behavior have been identified.

b. Experiment Objectives

The objectives of experiments in the rate-adaptive communications area are to develop and evaluate techniques for maintaining network performance and stability in the face of changing conditions. Schemes which provide graceful degradation of the quality of service as network loading approaches saturation are to be developed and tested. There will be a focus of attention on voice communication because of the variable-rate nature of the voice source. However, data flow-control techniques will also be tested. Of particular concern is whether the ARPANET class of flow-control techniques, which have been tested in a data-only environment, will perform effectively in an integrated voice/data network.

c. Experimental Approach and Requirements

Traffic emulation will be a primary tool in the testing of rate-adaptive techniques, since variable-rate voice coders are still in the developmental stages. Voice traffic characteristic of a large number of embedded coding vocoders will be emulated. For experiments on the satellite subnet, the priority-oriented mechanisms built into PODA will be used as a means for discarding lower-priority packets as loading increases. End-to-end rate-control mechanisms will be emulated and tested. To evaluate the effects on speech quality, one or two speech terminals capable of real embedded coding operation will be implemented in flexible high-speed processors. The speech from these terminals will be combined with emulated traffic, which will include other voice traffic as well as data traffic. The effect on voice performance of a sudden large demand for data capacity (e.g., a high-priority file transfer) will be evaluated. The performance of the highly dynamic embedded coding approach will be compared against simpler approaches such as rate control at dial-up.

d. Performance Measures

The primary performance measure for voice is the speech rate and associated quality achieved for the community of users as a function of network conditions. Stability is also an important performance measure. When many users try to respond to network conditions by raising and lowering their rates, network instabilities can result if proper controls are not applied. The proper trade-off between adaptation rate and stability must be made. For data traffic, the usual measures of delay, buffering requirements, and reliability are applicable. The network-wide implications of rate adaptation must be examined. Questions include an evaluation of the overall benefits to voice and data users as measured against the increased cost and complexity of the application of rate-adaptive techniques.

5. Routing

a. Introduction and Background

Previously, routing algorithms have been developed either primarily for voice traffic (the telephone network) or primarily for data traffic (the ARPANET). In the design of routing algorithms, satellite channels have generally been utilized as a cable in the sky rather than as a demand-assigned medium for flexible broadcast connectivity. Future integrated networks will include large volumes of both voice and data traffic, with flexible satellite connectivity in addition to terrestrial links. Routing algorithms should be matched to this environment, and the wideband network test bed provides an excellent facility for the development and testing of such algorithms.

b. Experiment Objectives

A major goal of the wideband project is the investigation of the interaction between the satellite and terrestrial components of a combined network. Of particular interest in this context are potential routing algorithms which involve choices between the utilization of satellite and terrestrial links. A mechanism must be provided for weighing the ability of the satellite to offer a path of fewer hops to a desired destination, against the disadvantage of the satellite round-trip delay. In addition, the routing algorithm must regularly operate in the context of a network with variable-capacity links, since the DAMA capability implies that capacity assignments on the satellite links will be adapted to changing traffic requirements. The design of a routing algorithm in a terrestrial/satellite network should also take into account the advantages afforded by the broadcast satellite medium for transmission of multi-addressed or conferenced data streams.

The interaction between voice and data traffic will also be of key concern, since the different delay and throughput requirements of these two types of traffic may call for different routing techniques. Voice traffic might be routed by a preselected fixed path or virtual circuit which has been determined to have sufficient capacity to accommodate the required bit rate, while interactive data might be routed on an independent packet-by-packet basis. The effect of data queue sizes in the network on the choice of voice virtual circuit routes through the network must also be considered.

c. Experimental Approach and Requirements

A prerequisite for routing experiments is a study to identify potential routing algorithms for the combined satellite/terrestrial network. Such a study should consider in detail the issues described above and identify important routing experiments. The applicability of existing routing algorithms to the wideband integrated environment should be considered. The design of the PSAT/ILRAN gateway should reflect the requirements for routing experiments.

d. Performance Measures

Routing performance measures include delay, throughput, cost, and reliability. Of particular interest will be the ability to choose between satellite and terrestrial paths in such a way that performance is optimized.

6. Conferencing

a. Introduction and Background

The capability for conferencing is an important requirement for military communications systems. In addition, voice conferencing provides an excellent vehicle for exercising a network with real traffic and for direct demonstration and observation of performance. Voice conferencing experiments and demonstrations have been an important facet of the DARPA Packet Speech Program and of the APSE Program. Conferencing experiments are expected to be of similar importance in the wideband experiment. The ability to handle greater numbers of voice users and to simultaneously communicate varying levels of real and simulated data traffic will add richness to conferencing experiments in the wideband network.

b. Experiment Objectives

Early conferencing experiments should focus on the transfer of APSE and ARPANET conferencing protocols to the wideband environment. These protocols generally implement control-signal switched conferencing, where the floor is switched from participant-to-participant by a chairman computer program which acts on the basis of participant request signals (push-buttons, touch tones, etc.). Of particular concern in these control-switched conference experiments will be the interaction between the conference controller program and PSAT IMP and gateway software. It is expected that conferencing on the satellite net will employ the stream mechanism of PODA, and experiments determining the capability and flexibility of this mechanism for conferencing purposes should be carried out as early as possible.

Another conferencing strategy to be tested is the voice-controlled technique where a participant's speech activity is monitored and he expresses his desire to talk simply by beginning to talk. Conferencing algorithms proposed for the World Wide Military Command and Control System (WWMCCS) utilize voice-activated switching in conjunction with a distributed control algorithm. Collisions on the satellite channel and momentary speech loss result when two or more users enter talkspurt within the same satellite round-trip time. These collisions result also in temporary loss of crypto sync on the channel. Programs to simulate these effects have been developed in a local test bed at Lincoln Laboratory. Experiments using this test bed are proceeding, and future evaluations of these algorithms will be carried out in an operational WWMCCS test bed. The experimental wideband network will allow the study and evaluation of these and similar conferencing techniques in an environment which is more realistic than a laboratory test facility, and more flexible than an operational conferencing system.

Consideration and simulation appropriate to the effects of security requirements will be an important aspect of these experiments. Conferencing will be carried out in the satellite subsystem. A later objective is the development, demonstration, and evaluation of an internetted conferencing capability for large numbers of conferees. This effort will serve as a follow-on to the current development of an internetted voice conferencing experimental facility involving the Atlantic Satellite Network and the ARPANET. The limited bandwidth in both these nets and the large delays in the ARPANET restrict the scope of internetted conferencing experiments in these nets. The wideband network will provide a more-substantial test bed for internetted conference experiments. The location of conference control and the interaction between the broadcast satellite network and the terrestrial net are subjects of particular concern.

Future military requirements may call for conferencing capabilities which include graphical as well as voice communication among participants. The wideband network will serve as an excellent facility for the development and demonstration of such a capability. Issues to be considered include the specification of required capabilities for voice/graphics terminals, the development of protocols for conference management, and the integration of voice and graphics data in the terminal access area.

c. Experimental Approach and Requirements

Conferencing represents a user application on the network. Therefore, conferencing experiments must be preceded by the establishment and verification of network operating capability. Internetted conferencing must be preceded by the development of gateways. For conferencing on the wideband satellite network, the availability of the stream PODA capability is of particular importance. Conferencing protocols developed for APSE presuppose a stream capability for maximum efficiency. Other obvious requirements for conferencing are voice terminals with suitable signaling mechanisms (push-buttons, touch tones, etc.) and network access capabilities. Finally, conferencing protocols and software must be designed and implemented.

The experimental approach will be to rely on transfer of existing (e.g., APSE) conferencing protocols to the wideband environment for initial conferencing experiments and demonstration of capabilities. Planning for more-advanced conferencing experiments such as voice/graphics conferencing should be a parallel effort.

d. Performance Measures

The wideband network test bed is not viewed as an appropriate facility for formal evaluation of conferencing strategies via human-factors experiments. Instead, the test bed will be used for feasibility demonstrations of different conferencing systems strategies. Evaluation of the success of particular strategies will be subjective but informal. Intelligibility of the speech, recognizability of talkers, and ability to gain the floor when required are important performance measures. Also of primary importance are ease of conference initiation and reliability, including the ability to maintain the conference even when some of the equipment (e.g., one of the PSAT IMPs) temporarily becomes disabled or loses communication.

7. Internetting

a. Introduction and Background

Internetting refers to communications between networks or subnets which operate with different communications protocols. Such communication is carried out with the aid of special processors called gateways, which reside at the internet boundaries and perform the functions necessary to transmit data from one network to another in usable form. Internetting is a subject of continuing research, as exemplified by the ongoing activities of the DARPA Internetting Program. In the packet-switching area, protocols for packet internetwork communication have been developed and implemented for experimental use in the ARPANET, Atlantic Satellite Net, and Packet Radio Net. The capability for intercommunication between different networks, whether circuit or packet switched, is a continuing requirement for operational military and civilian communications.

In the early phases of the wideband program, a limited form of internetting will serve as a convenient means for transferring central, monitoring, and measurements information between ARPANET hosts and PSAT IMP fake hosts. Later, experiments directed at internetting problems will become a focus of activity as the satellite and terrestrial networks become fully operational and development of gateways is initiated. Important problem areas to be addressed in these internetting experiments include circuit/packet interoperability, routing in combined satellite/terrestrial networks, and conferencing.

b. Experiment Objectives

The development and testing of a circuit-to-packet gateway for voice, including packetization/depacketization at the gateway, is an important experiment objective. Important issues to be investigated include requirements for gateway processing power, the effect of lost packets, and implications for privacy/security. The circuit/packet interoperability problem will arise specifically in the wideband test bed in experiments where voice is circuit-switched in the ILRANs and must traverse the packet-switched satellite net. The gateway development necessary to accommodate such communication will serve as a model for the later establishment of internet connections between the experimental network and operational circuit-switched voice networks such as AUTOVON and AUTOSEVOCOM II.

c. Experimental Approach and Requirements

Initial internet experiments will utilize an ARPANET/PSAT gateway, assuming that a decision is made to implement such a gateway. Hosts on the ARPANET will be used as a source for real data traffic (file transfers, etc.) to be transmitted across the wideband satellite net. The ARPANET speech capability can be adapted for use in internetted speech experiments.

An essential requirement for wideband internetting experiments is careful design of a wideband gateway between the satellite and terrestrial networks. The gateway must be flexible enough and have sufficient processing power to accommodate varied experiments with voice packetization/depacketization, satellite/terrestrial routing algorithms, and internet conferencing strategies. Study efforts directed at planning such experiments should begin as early as possible so that the required gateway functions can be specified.

d. Performance Measures

The effectiveness of the circuit/packet voice gateway will be determined on the basis of the ability to maintain voice communication with minimal added delay in passing through the gateway. This task as well as other internetting functions should be accomplished with as little cost as possible in terms of gateway processing power, memory requirements, and extra overhead added to the transmitted bit stream.

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This report documents work performed during FY 1978 on the DCA-sponsored Network Speech Processing Program. Three areas of work are reported: (1) a voice/data integration study investigating the effectiveness of combined circuit and packet multiplexing techniques; (2) a study of Demand-Assignment Multiple Access (DAMA) schemes for future integrated satellite networks; and (3) planning of experiments for an Experimental Integrated Switched Network (EISN) test bed.

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